# PATENT COOPERATION TREATY

# PCT

## INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

| Applicant's or agent's file reference AJR/37836 | **FOR FURTHER ACTION** see Notification of Transmittal of International Search Report (Form PCT/ISA/220) as well as, where applicable, item 5 below. | |
|---|---|---|
| International application No. PCT/GB 97/ 02159 | International filing date *(day/month/year)* 08/08/1997 | (Earliest) Priority Date *(day/month/year)* 09/08/1996 |
| Applicant KEMP, Michael Joseph | | |

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 2 sheets.

[X] It is also accompanied by a copy of each prior art document cited in this report.

1. [ ] **Certain claims were found unsearchable** (see Box I).

2. [ ] **Unity of invention is lacking** (see Box II).

3. [ ] The international application contains disclosure of a **nucleotide and/or amino acid sequence listing** and the international search was carried out on the basis of the sequence listing
   - [ ] filed with the international application.
   - [ ] furnished by the applicant separately from the international application,
     - [ ] but not accompanied by a statement to the effect that it did not include matter going beyond the disclosure in the international application as filed.
   - [ ] Transcribed by this Authority

4. With regard to the **title**,
   - [X] the text is approved as submitted by the applicant
   - [ ] the text has been established by this Authority to read as follows:

5. With regard to the **abstract**,
   - [X] the text is approved as submitted by the applicant
   - [ ] the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this International Search Report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is:

   Figure No. 8
   - [ ] as suggested by the applicant.
   - [X] because the applicant failed to suggest a figure.
   - [ ] because this figure better characterizes the invention.
   - [ ] None of the figures.

# INTERNATIONAL SEARCH REPORT

International Application No
PCT/GB 97/02159

**A. CLASSIFICATION OF SUBJECT MATTER**
IPC 6 G10K15/02

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 G10K

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

| Category ° | Citation of document, with indication, where appropriate, of the relevant passages | Relevant to claim No. |
|---|---|---|
| A | US 5 544 249 A (OPITZ MARTIN) 6 August 1996<br>see claim 1<br>----- | 1,3,4,<br>10,12,13 |

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance
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"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
"&" document member of the same patent family

1 Date of the actual completion of the international search: 15 December 1997

Date of mailing of the international search report: 23/12/1997

Name and mailing address of the ISA
European Patent Office, P.B. 5818 Patentlaan 2
NL - 2280 HV Rijswijk
Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,
Fax: (+31-70) 340-3016

Authorized officer
Anderson, A

# INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/GB 97/02159

| Patent document cited in search report | Publication date | Patent family member(s) | Publication date |
|---|---|---|---|
| US 5544249 A | 06-08-96 | DE 4328620 C | 19-01-95 |
| | | EP 0641143 A | 01-03-95 |
| | | JP 7087589 A | 31-03-95 |

# Best Available Copy

PCT/GB97/02159

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From the INTERNATIONAL BUREAU

## PCT

**NOTIFICATION OF ELECTION**

(PCT Rule 61.2)

To:

United States Patent and Trademark Office
(Box PCT)
Crystal Plaza 2
Washington, DC 20231
ETATS-UNIS D'AMERIQUE

in its capacity as elected Office

| | |
|---|---|
| **Date of mailing** (day/month/year)<br>19 March 1998 (19.03.98) | |
| **International application No.**<br>PCT/GB97/02159 | **Applicant's or agent's file reference**<br>AJR/37836 |
| **International filing date** (day/month/year)<br>08 August 1997 (08.08.97) | **Priority date** (day/month/year)<br>09 August 1996 (09.08.96) |
| **Applicant**<br>KEMP, Michael, Joseph | |

1. The designated Office is hereby notified of its election made:

   [X] in the demand filed with the International Preliminary Examining Authority on:

   27 February 1998 (27.02.98)

   [ ] in a notice effecting later election filed with the International Bureau on:

2. The election [X] was [ ] was not made before the expiration of 19 months from the priority date or, where Rule 32 applies, within the time limit under Rule 32.2(b).

| | |
|---|---|
| **The International Bureau of WIPO**<br>**34, chemin des Colombettes**<br>**1211 Geneva 20, Switzerland**<br>Facsimile No.: (41-22) 740.14.35 | Authorized officer<br>Aino Metcalfe<br>Telephone No.: (41-22) 338.83.38 |

COPY

# PCT

## REQUEST

The undersigned requests that the present international application be processed according to the Patent Cooperation Treaty.

| For receiving Office use only |
|---|
| International Application No. |
| International Filing Date |
| Name of receiving Office and "PCT International Application" |

Applicant's or agent's file reference *(if desired) (12 characters maximum)* AJR/37836

**Box No. I** TITLE OF INVENTION

AUDIO EFFECTS SYNTHESIZER WITH OR WITHOUT ANALYSER

**Box No. II** APPLICANT

Name and address: *(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country.)*

**KEMP,** MICHAEL J
of Casa Lucia
Vale Formosilho
S. Marcos da Serra
8375 S.B. de Messines
PORTUGAL

[X] This person is also inventor.

Telephone No. 082 361448

Facsimile No. 082 361448

Teleprinter No.

State *(i.e. country)* of nationality: UK

State *(i.e. country)* of residence: PT

This person is applicant for the purposes of: [X] all designated States [ ] all designated States except the United States of America [ ] the United States of America only [ ] the States indicated in the Supplemental Box

**Box No. III** FURTHER APPLICANT(S) AND/OR (FURTHER) INVENTOR(S)

Name and address: *(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country.)*

This person is:

[ ] applicant only

[ ] applicant and inventor

[ ] inventor only *(If this check-box is marked, do not fill in below.)*

State *(i.e. country)* of nationality:

State *(i.e. country)* of residence:

This person is applicant for the purposes of: [ ] all designated States [ ] all designated States except the United States of America [ ] the United States of America only [ ] the States indicated in the Supplemental Box

[ ] Further applicants and/or (further) inventors are indicated on a continuation sheet.

**Box No. IV** AGENT OR COMMON REPRESENTATIVE; OR ADDRESS FOR CORRESPONDENCE

The person identified below is hereby/has been appointed to act on behalf of the applicant(s) before the competent International Authorities as: [X] agent [ ] common representative

Name and address: *(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country.)*

**ROBSON,** AIDAN JOHN
Reddie & Grose
16 Theobalds Road
London WC1X 8PL
UNITED KINGDOM

Telephone No. 0044 171 242-0901

Facsimile No. 0044 171 242-3290

Teleprinter No.

[ ] Mark this check-box where no agent or common representative is/has been appointed and the space above is used instead to indicate a special address to which correspondence should be sent.

**Box No. V** **DESIGNATION OF STATES**

The following designations are hereby made under Rule 4.9(a) *(mark the applicable check-boxes; at least one must be marked)*:

**Regional Patent**

☒ **AP** **ARIPO Patent:** **GH** Ghana, **KE** Kenya, **LS** Lesotho, **MW** Malawi, **SD** Sudan, **SZ** Swaziland, **UG** Uganda, **ZW** Zimbabwe, and any other State which is a Contracting State of the Harare Protocol and of the PCT

☒ **EA** **Eurasian Patent:** **AM** Armenia, **AZ** Azerbaijan, **BY** Belarus, **KG** Kyrgyzstan, **KZ** Kazakstan, **MD** Republic of Moldova, **RU** Russian Federation, **TJ** Tajikistan, **TM** Turkmenistan, and any other State which is a Contracting State of the Eurasian Patent Convention and of the PCT

☒ **EP** **European Patent:** **AT** Austria, **BE** Belgium, **CH and LI** Switzerland and Liechtenstein, **DE** Germany, **DK** Denmark, **ES** Spain, **FI** Finland, **FR** France, **GB** United Kingdom, **GR** Greece, **IE** Ireland, **IT** Italy, **LU** Luxembourg, **MC** Monaco, **NL** Netherlands, **PT** Portugal, **SE** Sweden, and any other State which is a Contracting State of the European Patent Convention and of the PCT

☒ **OA** **OAPI Patent:** **BF** Burkina Faso, **BJ** Benin, **CF** Central African Republic, **CG** Congo, **CI** Côte d'Ivoire, **CM** Cameroon, **GA** Gabon, **GN** Guinea, **ML** Mali, **MR** Mauritania, **NE** Niger, **SN** Senegal, **TD** Chad, **TG** Togo, and any other State which is a member State of OAPI and a Contracting State of the PCT *(if other kind of protection or treatment desired, specify on dotted line)* ..........

**National Patent** *(if other kind of protection or treatment desired, specify on dotted line)*:

- ☒ **AL** Albania ..........
- ☒ **AM** Armenia ..........
- ☒ **AT** Austria ..........
- ☒ **AU** Australia ..........
- ☒ **AZ** Azerbaijan
- ☒ **BA** Bosnia and Herzegovina ..........
- ☒ **BB** Barbados
- ☒ **BG** Bulgaria ..........
- ☒ **BR** Brazil ..........
- ☒ **BY** Belarus
- ☒ **CA** Canada
- ☒ **CH and LI** Switzerland and Liechtenstein
- ☒ **CN** China ..........
- ☒ **CU** Cuba ..........
- ☒ **CZ** Czech Republic ..........
- ☒ **DE** Germany ..........
- ☒ **DK** Denmark ..........
- ☒ **EE** Estonia ..........
- ☒ **ES** Spain ..........
- ☒ **FI** Finland ..........
- ☒ **GB** United Kingdom
- ☒ **GE** Georgia ..........
- ☒ **GH** Ghana ..........
- ☒ **HU** Hungary ..........
- ☒ **IL** Israel ..........
- ☒ **IS** Iceland
- ☒ **JP** Japan ..........
- ☒ **KE** Kenya ..........
- ☒ **KG** Kyrgyzstan ..........
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- ☒ **KR** Republic of Korea ..........
- ☒ **KZ** Kazakstan ..........
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- ☒ **LK** Sri Lanka
- ☒ **LR** Liberia
- ☒ **LS** Lesotho ..........
- ☒ **LT** Lithuania
- ☒ **LU** Luxembourg ..........
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- ☒ **MG** Madagascar ..........
- ☒ **MK** The former Yugoslav Republic of Macedonia ..........
- ☒ **MN** Mongolia
- ☒ **MW** Malawi ..........
- ☒ **MX** Mexico ..........
- ☒ **NO** Norway
- ☒ **NZ** New Zealand ..........
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- ☒ **TM** Turkmenistan ..........
- ☒ **TR** Turkey ..........
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- ☒ **UG** Uganda ..........
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- ☒ **VN** Viet Nam ..........
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- ☒ **ZW** Zimbabwe

Check-boxes reserved for designating States (for the purposes of a national patent) which have become party to the PCT after issuance of this sheet:

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- ☐ ..........
- ☐ ..........
- ☐ ..........
- ☐ ..........

**In addition to the designations made above, the applicant also makes under Rule 4.9(b) all designations which would be permitted** under the PCT except the designation(s) of ____________

The applicant declares that those additional designations are subject to confirmation and that any designation which is not confirmed before the expiration of 15 months from the priority date is to be regarded as withdrawn by the applicant at the expiration of that time limit. *(Confirmation of a designation consists of the filing of a notice specifying that designation and the payment of the designation and confirmation fees. Confirmation must reach the receiving Office within the 15-month time limit.)*

**Box No. VI PRIORITY CLAIM** — Further priority claims are indicated in the Supplemental Box ☐

The priority of the following earlier application(s) is hereby claimed:

| Country *(in which, or for which, the application was filed)* | Filing Date *(day/month/year)* | Application No. | Office of filing *(only for regional or international application)* |
|---|---|---|---|
| item (1) UNITED KINGDOM | 9 August 1996 09/08/1996 | 9616755.6 | |
| item (2) | | | |
| item (3) | | | |

*Mark the following check-box if the certified copy of the earlier application is to be issued by the Office which for the purposes of the present international application is the receiving Office (a fee may be required):*

[X] The receiving Office is hereby requested to prepare and transmit to the International Bureau a certified copy of the earlier application(s) identified above as item(s) : (1)

**Box No. VII INTERNATIONAL SEARCHING AUTHORITY**

**Choice of International Searching Authority (ISA)** *(If two or more International Searching Authorities are competent to carry out the international search, indicate the Authority chosen; the two-letter code may be used):* ISA /

**Earlier search** *Fill in where a search (international, international-type or other) by the International Searching Authority has already been carried out or requested and the Authority is now requested to base the international search, to the extent possible, on the results of that earlier search. Identify such search or request either by reference to the relevant application (or the translation thereof) or by reference to the search request:*

Country (or regional Office): Date *(day/month/year)*: Number:

**Box No. VIII CHECK LIST**

This international application contains the following number of sheets:

| | | |
|---|---|---|
| 1. request | : 3 | sheets |
| 2. description | : 30 | sheets |
| 3. claims | : 4 | sheets |
| 4. abstract | : 1 | sheets |
| 5. drawings | : 20 | sheets |
| Total | : 58 | sheets |

This international application is accompanied by the item(s) marked below:

1. ☐ separate signed power of attorney
2. ☐ copy of general power of attorney
3. ☐ statement explaining lack of signature
4. ☐ priority document(s) *identified in Box No. VI as item(s):*
5. [X] fee calculation sheet
6. ☐ separate indications concerning deposited microorganisms
7. ☐ nucleotide and/or amino acid sequence listing (diskette)
8. ☐ other *(specify)*:

Figure No. ______ of the drawings (if any) should accompany the abstract when it is published.

**Box No. IX SIGNATURE OF APPLICANT OR AGENT**

*Next to each signature, indicate the name of the person signing and the capacity in which the person signs (if such capacity is not obvious from reading the request).*

KEMP, MICHAEL J

ROBSON, Aidan John
Authorised Representative

For receiving Office use only

1. Date of actual receipt of the purported international application:
3. Corrected date of actual receipt due to later but timely received papers or drawings completing the purported international application:
4. Date of timely receipt of the required corrections under PCT Article 11(2):
5. International Searching Authority specified by the applicant: ISA /
6. ☐ Transmittal of search copy delayed until search fee is paid

2. Drawings:
☐ received:
☐ not received:

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Date of receipt of the record copy by the International Bureau:

*This sheet is not part of and does not count as a sheet of the international application.*

# PCT

## FEE CALCULATION SHEET

**Annex to the Request**

For receiving Office use only

International application No.

Date stamp of the receiving Office

Applicant's or agent's file reference: AJR/37836

Applicant

KEMP, Michael J

### CALCULATION OF PRESCRIBED FEES

1. TRANSMITTAL FEE . . . . . . . . . . . . . . . . . . . . . . [ ] T
2. SEARCH FEE . . . . . . . . . . . . . . . . . . . . . . . . [ ] S

   International search to be carried out by ____________

   *(If two or more International Searching Authorities are competent in relation to the international application, indicate the name of the Authority which is chosen to carry out the international search.)*

3. INTERNATIONAL FEE

   **Basic Fee**

   The international application contains ____ sheets.

   first 30 sheets . . . . . . . . . . . . . . [ ] $b_1$

   ________ x ________ = [ ] $b_2$

   remaining sheets — additional amount

   Add amounts entered at $b_1$ and $b_2$ and enter total at B . . . . [ ] B

   **Designation Fees**

   The international application contains ____ designations.

   ________ x ________ = [ ] D

   number of designation fees payable *(maximum 11)* — amount of designation fee

   Add amounts entered at B and D and enter total at I . . . . . . . . . . [ ] I

   *(Applicants from certain States are entitled to a reduction of 75% of the international fee. Where the applicant is (or all applicants are) so entitled, the total to be entered at I is 25% of the sum of the amounts entered at B and D.)*

4. FEE FOR PRIORITY DOCUMENT . . . . . . . . . . . . . . . . [ ] P
5. TOTAL FEES PAYABLE

   Add amounts entered at T, S, I and P, and enter total in the TOTAL box . . [ ] TOTAL

☐ The designation fees are not paid at this time.

### MODE OF PAYMENT

- ☐ authorization to charge deposit account (see below)
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International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

| | | |
|---|---|---|
| (51) International Patent Classification 6 : G10K 15/02 | A1 | (11) International Publication Number: WO 98/07141<br>(43) International Publication Date: 19 February 1998 (19.02.98) |

(21) International Application Number: PCT/GB97/02159

(22) International Filing Date: 8 August 1997 (08.08.97)

(30) Priority Data:
9616755.6 9 August 1996 (09.08.96) GB

(71)(72) Applicant and Inventor: KEMP, Michael, Joseph [GB/PT]; Casa Lucia, Vale Formosilho, S. Marcos da Serra, P-8375 S.B. de Messines (PT).

(74) Agent: ROBSON, Aidan, John; Reddie & Grose, 16 Theobalds Road, London WC1X 8PL (GB).

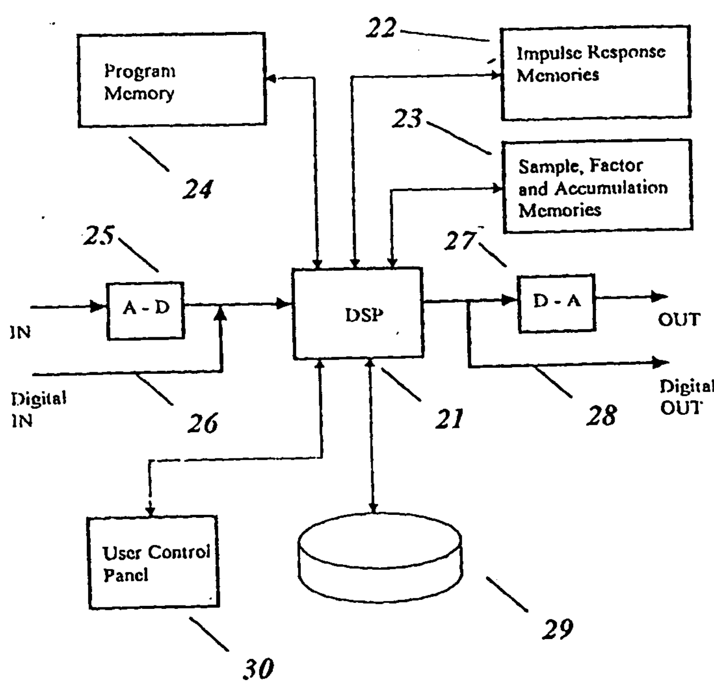
(81) Designated States: AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CU, CZ, DE, DK, EE, ES, FI, GB, GE, GH, HU, IL, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, UA, UG, US, UZ, VN, YU, ZW, ARIPO patent (GH, KE, LS, MW, SD, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, ML, MR, NE, SN, TD, TG).

**Published**
*With international search report.*
*Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.*

(54) Title: AUDIO EFFECTS SYNTHESIZER WITH OR WITHOUT ANALYSER

(57) Abstract

The method is provided for simulating an audio effect processor. At least two impulse responses are stored representing the audio processor for input signals having different characteristics. The same characteristic of the input signal is then repeatedly assessed and at least one of the impulse responses selected to apply to the input signal in dependence on the result of the assessment. The selected impulse response is then applied to the input signal to derive an output signal. This process continues throughout the duration of the input signal thereby taking account of time dependent changes in the assessed characteristic.



*FOR THE PURPOSES OF INFORMATION ONLY*

Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

| | |
|---|---|
| AL | Albania |
| AM | Armenia |
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| AZ | Azerbaijan |
| BA | Bosnia and Herzegovina |
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| BY | Belarus |
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| HU | Hungary |
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| IL | Israel |
| IS | Iceland |
| IT | Italy |
| JP | Japan |
| KE | Kenya |
| KG | Kyrgyzstan |
| KP | Democratic People's Republic of Korea |
| KR | Republic of Korea |
| KZ | Kazakstan |
| LC | Saint Lucia |
| LI | Liechtenstein |
| LK | Sri Lanka |
| LR | Liberia |
| LS | Lesotho |
| LT | Lithuania |
| LU | Luxembourg |
| LV | Latvia |
| MC | Monaco |
| MD | Republic of Moldova |
| MG | Madagascar |
| MK | The former Yugoslav Republic of Macedonia |
| ML | Mali |
| MN | Mongolia |
| MR | Mauritania |
| MW | Malawi |
| MX | Mexico |
| NE | Niger |
| NL | Netherlands |
| NO | Norway |
| NZ | New Zealand |
| PL | Poland |
| PT | Portugal |
| RO | Romania |
| RU | Russian Federation |
| SD | Sudan |
| SE | Sweden |
| SG | Singapore |
| SI | Slovenia |
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| US | United States of America |
| UZ | Uzbekistan |
| VN | Viet Nam |
| YU | Yugoslavia |
| ZW | Zimbabwe |

# INTERNATIONAL SEARCH REPORT

Intr 'ional Application No
PCT/GB 97/02159

**A. CLASSIFICATION OF SUBJECT MATTER**
IPC 6 G10K15/02

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 G10K

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

| Category ° | Citation of document, with indication, where appropriate, of the relevant passages | Relevant to claim No. |
|---|---|---|
| A | US 5 544 249 A (OPITZ MARTIN) 6 August 1996<br>see claim 1<br>----- | 1,3,4,<br>10,12,13 |

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier document but published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
"&" document member of the same patent family

| Date of the actual completion of the international search | Date of mailing of the international search report |
|---|---|
| 15 December 1997 | 23/12/1997 |

| Name and mailing address of the ISA | Authorized officer |
|---|---|
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(71)(72) Applicant and Inventor: KEMP, Michael, Joseph [GB/PT]; Casa Lucia, Vale Formosilho, S. Marcos da Serra, P-8375 S.B. de Messines (PT).

(74) Agent: ROBSON, Aidan, John; Reddie & Grose, 16 Theobalds Road, London WC1X 8PL (GB).

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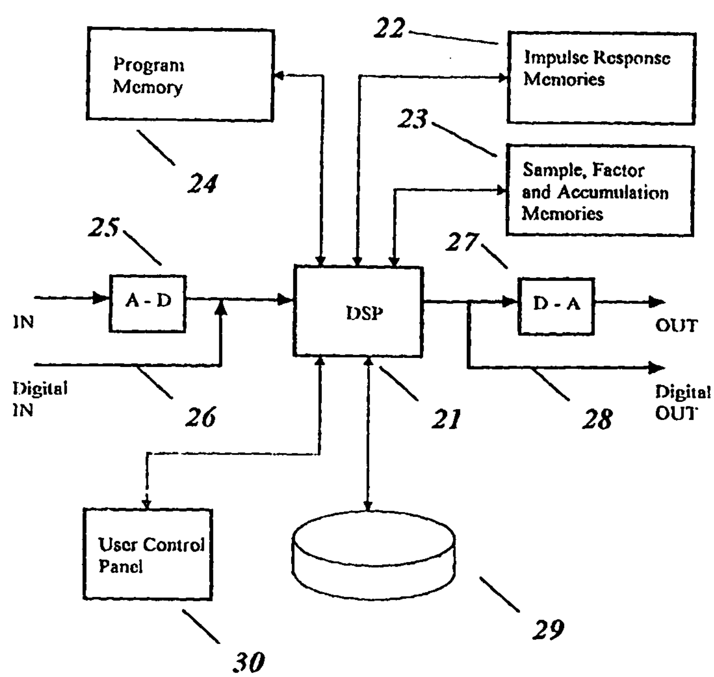
(54) Title: AUDIO EFFECTS SYNTHESIZER WITH OR WITHOUT ANALYSER

(57) Abstract

The method is provided for simulating an audio effect processor. At least two impulse responses are stored representing the audio processor for input signals having different characteristics. The same characteristic of the input signal is then repeatedly assessed and at least one of the impulse responses selected to apply to the input signal in dependence on the result of the assessment. The selected impulse response is then applied to the input signal to derive an output signal. This process continues throughout the duration of the input signal thereby taking account of time dependent changes in the assessed characteristic.



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## Audio Effects Synthesizer with or without Analyser

## Introduction

In audio recording for music or film it is often desired to pass an audio signal through an effect unit to alter the sound in a desirable way, for example, in film work a recording may be made to sound as if it were coming through a telephone, from a distance or in a room with characteristic sound quality even though the original sound was recorded in a dead acoustic of a studio. In music work more severe distortions may be required, for example passing the signal through a guitar amplifier and speaker which is allowed to distort and back into a microphone, or through an analogue recording cycle onto and back from magnetic tape which is often considered to add a desirable sound quality.

Many devices exist to process signals in these ways, some specific to individual effects and some programmable to generate a range of effects on demand. The purpose of this invention is to allow the simulation of a large variety of such effects and further to allow existing effects to be analysed and the characteristics of the effect to be stored and simulated on demand.

## List of figures

The invention is described by means of reference to the attached figures which are described in detail after the following summary explanation.

**Fig 1** shows the process of analysing an existing effect unit by means of applying an impulse and recording its impulse response.

**Fig 2** shows the application of an input sound stream to generate a processed output stream by convolution with the sampled impulse response.

**Fig 3** shows the application of impulses of different magnitudes to an effect unit to obtain more than one impulse response appropriate to different impulse amplitudes.

**Fig 4** shows the application of an input stream to generate a processed output stream by modifying the convolution so that a different impulse response may be applied to different input samples - in this case depending on amplitude of the input sample

compared with a threshold shown chain-dotted.

**Fig 5** shows a further refinement where an input sample between two thresholds is applied proportionately to the two impulse responses appropriate to the thresholds on either side of the input sample.

**Fig 6** shows an alternative step pulse that may be applied in the analysis process.

**Fig 7** shows the derivation of the impulse response from the step response by means of a sample shift and a subtraction.

**Fig 8** shows an arrangement of DSP and memory which can implement the steps of (i) analysing a device by means of generating impulses, storing the responses returned from an effect under analysis and performing various 'tidying up' algorithms as described below to create the stored impulse responses, (ii) reading an input sample and generating the sample, factor and address data for storage in memory as shown in fig 10, and (iii) executing the algorithm of fig 12 to generate each output sample after each input sample has been read in, compared with impulse response thresholds, and stored. A fixed or removable disc drive may also be provided for program storage, long-term storage of response data and exchange of data between machines.

**Fig 9** shows part of one method of implementing the simulation process wherein an input sample is analysed once to determine two impulse responses to be applied to it, where the start address of the impulse response stream in memory of the lower response appropriate to this sample is stored, and where the sample is divided proportionally as determined by the proximity of the sample amplitude to the two impulse response amplitudes ready for subsequent processing.

**Fig 10** shows the algorithm to be applied to derive the values to be stored in fig 9,

**Fig 11** shows the arrangement in memory after the most recent input sample has been divided and placed in memory at $F_1(0)$ and $F_2(0)$ together with the selected address of the lower of the two appropriate impulse responses stored at A(0). The previous samples derived values are stored at $F_1(1)$,$F_2(1)$, $F_1(2)$, $F_2(2)$ etc together with their associated A pointers for sufficient previous samples to at least equal the length of the impulse responses used in the simulation.

**Fig 12** shows the algorithm used to calculate an output sample from the data stored

in memory in fig 11.

**Fig 13** shows one possible multiple processor implementation wherein DSP1 is used first to analyse an effect and generate the sampled impulse responses, then is used during the simulation phase to generate the sample and factor memory entries. This memory is segmented into a number of areas each of which is accessible to its own DSP (2,3,4...) which can thus calculate part contributions to each output sample. These part sums are then fed back to DSP1 to be summed to generate the total output sample and fed to the output.

**Fig 14** shows an alternative way to implement the simulation algorithm where the heavily repeated inner loop of the convolution algorithm is simplified for maximum speed of execution, requiring a simple multiply of each element of the impulse response buffer and accumulation into each element of the output sample buffer.

**Fig 15** shows the digital signal of an appropriate analysis tone which may be applied to a device under test remotely from the tone generating and analysis device by, for example, recording the test signal and applying it to the device under test and recording the impulse responses resulting for later analysis

**Fig 16** shows an alternative test signal which may be used when the device under test is available at the same time as the generator and analyser device

**Fig 17** shows a flow diagram of a process to generate the test pulses of fig 16 and record the impulse responses during analysis.

**Fig 18** shows a flow diagram of an alternative process to generate impulse test pulses rather than a stepped test pulse and to record the impulse responses during analysis.

**Fig 19** shows a noise removal strategy where impulse responses derived from lower amplitude impulses may be selectively replaced by impulse responses from higher amplitude impulses in areas where the mean amplitude of the impulse response falls below a threshold representing the approach to a noise floor which would impair the simulation process.

**Fig 20** shows the process of removing jitter from a signal recovered from a device or process under test, for example where there is randomised delay in the device (e.g. wow and flutter) or where the sampling process clock is not locked digitally to the analysis tone

generator.

**Fig 21** shows the steps required to select between impulse responses based on the envelope of the incoming signal rather than instantaneous amplitude.

## Analysis and Simulation of linear systems

It is known that the transfer characteristic of a linear audio processor can be characterised by its impulse response. A single pulse can be passed through an effect unit and the resulting signal which emerges can be recorded as a sequence of digital samples. The effect can then be simulated in the digital domain by convolving a digital input stream with this impulse response to produce a digital output stream which matches that which would have emerged from the sampled effect unit. The impulse response can be stored for recall later. This is illustrated in figure 1 where an impulse *T* is applied via a D/A converter 1 to produce an analogue impulse 2 which is fed into effect unit 3. The output impulse response waveform 4 is fed via digital to analogue converter 5 and the resulting impulse response *R* is measured and stored.. Fig. 2 shows how the resulting impulse response *R* is used to calculate an output stream *O* from input stream *I*. The most recent sample received and output is suffixed 0, with progressively older samples suffixed 1,2,3 etc. Output sample *O*(0) is derived by taking the most recent input sample *I*(0) and multiplying this by the first sample of response *R* (*R*(0)shown at 7), summed (or *accumulated*) with the product of *I*(1) and the next older impulse sample (*R*(1) shown at 8) and so on until the oldest input sample required *I*(6) is multiplied by *R*(6) (shown at 10) is accumulated to make the latest output sample *O*(0). Thus the input stream of data *I* representing an input audio signal is convolved with the single impulse response *R* to produce each sample in output stream *O*. Although 6 samples are referred to here for the length of the impulse responses, this is for clarity only and in practice many more samples are used. Although multiple output samples are shown, in fact it is not necessary to store these values as a new output sample is derived when each new input sample is received and may be fed directly to the output.

Where the effect unit to be analysed already has digital input and/or output the D/A

(1) or the A/D (5) may not be required as the digital signals can simply be fed to or fed back from the effect unit..

## Extension to non-linear systems

Many effects including some of those mentioned above are non linear in nature and the response of a signal path depends on the level of signal passing through the unit. According to this invention it is possible to analyse such an effects unit by applying a number of different impulses of different amplitude and to store a different resulting impulse response from each exciting impulse. This is illustrated in fig 3 for two different pulse amplitudes at fig 3(a) and fig 3(b). Figure 3(a) duplicates the process shown in fig 1, using a sample pulse *T* of maximum amplitude to determine the response of the system under maximum amplitude conditions. Figure 3(b) duplicates the test but using a lower amplitude impulse *T'*, for example half the amplitude of the pulse in fig 3(a). The resulting impulse response is shown at *R*. This is then increased in amplitude to produce the response at *R'* by multiplying each sample by the ratio of the maximum sample amplitude at *T* over the lower sample amplitude at *T'*. This process is known as normalisation..

In practice, to obtain a good analysis of the non-linear response of the system, a number of different impulse levels are applied and a set of impulse responses (normalised to maximum amplitude) are obtained. Typically a set of 128 or 256 impulse responses are used using an equally spaced set of sample impulses from the maximum level down to 1/128 (or 1/256 in the latter case) of the maximum level. In the case of 128 steps being used the response of the system is thus determined for signals from the maximum level down to 42dB below this, at which point most effects have become linear.

After obtaining the set of impulse responses it is possible to simulate the non-linear effect. When simulating the effect it is necessary to examine each input sample and depending on the magnitude of the sample to use the appropriate impulse response in the convolution. This is shown in figure 4 for the case where the set of impulse responses uses just the two responses obtained in fig 3 and by comparison with fig. 2. Each input sample

(at $I$) needed to make up the output sample is compared against the threshold determined by the magnitude of the lower impulse of figure 3b, shown as chain-dotted line 11. If the magnitude input sample exceeds this threshold (i.e $I(3)$, $I(4)$ and $I(5)$), the impulse response of the higher amplitude pulse (shown replicated for each input sample considered at 12) is used in the convolution. If the magnitude of the input sample is below the threshold (i.e. $I(0)$, $I(1)$, $I(2)$, $I(6)$) the impulse response of the lower amplitude impulse (13) is used in the convolution calculation. Once again all contributing products of input samples and appropriate impulse response are summed to generate the desired next output value $O(0)$

This process can be extended to use the impulse responses of any number of different impulse amplitudes by comparing the input sample against a number of thresholds. In the example where there are 128 equally spaced test impulses used to derive the impulse response set, the appropriate response to use for any sample can be simply obtained by truncation of the magnitude of the sample to 7 bits (equivalent to 128 levels). The magnitude means that the sign of the sample value is removed to determine solely its amplitude.

In fact it can be seen that the number of calculations required to generate an output sample is increased only by the need to make a decision for each input sample. The decision needs only to be taken once for each input sample (regardless of how many times this sample needs to be used to calculate subsequent output samples) so in fact represents only a small increase in computational complexity. This is shown in the later detailed description of the process of simulation. Thus it is possible to use a large number of different impulse responses representing, say, 128 different sample levels without increasing the number of calculations by anything like the number of levels used.

Whilst the principle implementations described here take a single impulse response at each level and disregards the sign of the input signal during simulation (using only the magnitude for determining which impulse response to use), it is possible to simulate

effects which have significant asymmetrical response by storing responses to both positive and negative going transitions, and applying the one appropriate to the sign of each input sample as well as magnitude.

## Improvement by linear interpolation of impulse responses

Whilst the above process provides a simulation of the sampled effect, an improvement in distortion characteristics can be made if desired at the expense of some increase in computational complexity by modifying the process so that instead of selecting between two different impulse responses at a given level, a cross-fade effect is used applying a proportion of the input sample to two impulse responses representing two adjacent impulse levels. This is shown in fig 5 where a sample (14) a quarter of the way between two sample thresholds (15, 16) is applied three-quarters to the impulse response representing the lower sample level (17) and one quarter to the impulse response representing the higher sample level (18). No calculation needs to be performed with any of the other impulse responses. The computational complexity has thus doubled over the simple case of fig 2 plus the additional computation to compute the ratio between the two levels. Although this represents more complexity than of the simple case of fig 2, it still represents an acceptable level of complexity to achieve the non-linear characteristic of many simulated effects, as once again this can be evaluated just once for each input sample.

## Switching between modes

In fact the simulator can be made to switch between the three cases of the simple linear simulator of fig 2, the non-linear simulator of fig 4 and the improved non-linear simulator of fig 5 according to the available computational power and according the length of the impulse responses used. This switching can be achieved by changing the stored program executed by the DSP processors used to implement the system.

## Reducing Noise in the sampled impulse response using an alternative sampling pulse

The analysis pulse of fig 1 generates an impulse response but the resulting impulse response may also contain noise. Low frequency noise tends to be correlated between adjacent samples and during the resulting simulation phase may lead to either large DC offsets or general low frequency noise on the resulting output.

Figure 6 shows that instead of the unit impulse test signal *T* of figure 1 the step pulse *ST* may be applied. The step response *SR* is thus obtained.

Figure 7 shows how to recover the unit impulse response required *R*. The step impulse response *SR* is shifted on by one sample to get *SR'* which is subtracted sample by sample from the response *SR* to yield the desired impulse response *R*. Thus any substantial correlation between samples is largely eliminated, and any DC offset (i.e. a constant bias found on all analysed samples) is totally removed. This can of course be calculated as the impulse response is sampled by storing the previous sample value $S_{n-1}$ and subtracting it from the current sample $S_n$ so the value $S_n - S_{n-1}$ is stored as the desired impulse response.

The desired response at the required number of different amplitudes can be found by using steps of a number of different sizes, as shown in fig 15 and described later.

## Implementing the Analysis and Simulation

The implementation of the analysis and simulating process will now be described by reference to figures 8 to 18. Figure 8 shows one arrangement using a stored program computer optimised for digital signal processing. Typically one or more digital signal processor (DSP) devices 21 are used. The DSP is attached to memory for impulse responses 22 and for digital audio sample, accumulation and control data 23, as well as program memory 24. These may in fact be part of a single general purpose memory array or for example the program memory 24 may be part of a separate array for higher performance. Audio input is provided either via analogue to digital converter 25 or via direct digital input 26 and audio output is fed via digital to analogue converter 27 and via a

direct digital output 28. A disk storage subsystem 29 is also connected and a user control panel and display 30 is provided to allow the user to initiate analysis, store or select stored impulse responses, and select simulation modes, as well as initiate editing of impulse responses as described later.

The arrangement of fig 8 may generate the analysis pulses, store and process the resultant impulse responses, and produce the simulation by loading the appropriate control program from disk or other storage medium.

One method of implementing the process of simulation will be described first.

Figure 9 shows the process of reading in samples to be processed. A number of impulse responses derived from the analysis phase are stored in arrays of memories shown at 31, 32 and 33. Three are shown but in practice any number may be used. These are identified by the address in memory of the first element of each response at 34, 35 and 36 and the memory array A can store these memory addresses, or pointers, represented by the arrows shown from memory elements of array A pointing to the appropriate impulse array. Each input sample arriving has one element of A reserved for it to denote the appropriate pair of impulse responses 31 - 33. The pointer addresses the lower of the two impulse responses (i.e. the impulse response derived from the lower magnitude analysis impulse) representing the threshold on or below the input sample, and the second impulse response is always the next one above representing the next higher threshold level of the input sample.

Memory arrays $F_1$ and $F_2$ store a pair of factors which are derived from the input sample and represent the input sample divided into two parts, one of which will be applied to the lower impulse response and one of which will be applied to the higher impulse response. The sum of these two factors is always the input sample value itself and the sample is divided and stored in elements of arrays $F_1$ and $F_2$ according to the proportion to be applied to each impulse response. Each input sample 37 therefore is divided in process

38 and loaded into the next free set of elements of the arrays A, $F_1$, and $F_2$. A pointer 39 is then incremented (to the left in this example) to point into the next set of elements for the next input sample when it arrives.

Figure 10 shows by means of a flow diagram the details of the process 38. The magnitude |S| of the input sample S is compared with the various thresholds $T_1$, $T_2$ etc representing the levels at which the impulse responses were sampled, to find the two thresholds $T_n$ on or below the sample magnitude, and $T_{n+1}$ above the sample magnitude, i.e. such that

$$T_n \leq |S| < T_{n+1}.$$

It should be noted that if the number of equally spaced levels is a power of 2 (e.g. 256) the threshold value $T_n$ can be determined by first removing the sign of the sample value then truncation to the number of bits appropriate to the power of 2, (e.g. 8).

The next step is to calculate the proportion by which the sample amplitude exceeds the threshold (shown as factor k), then divide the sample in this proportion to place into arrays $F_1$ and $F_2$.

The input pointer is then advanced ready for the next sample. The array stored will be used for calculating each output samples up to the length of the impulse responses, so after a number of output samples the values just calculated will no longer be required. Standard techniques may be applied to implement a 'circular buffer' where the pointer can be wrapped back to the start after this many samples. thus limiting the size of the arrays. These techniques are well known and do not need to be described further here.

Figure 11 thus shows the layout of data in memory after a number of samples have been read in and processed to calculate output samples (ignoring any issues relating to circular buffers). In this example and in the process shown in fig 12 the parenthesised

suffix (0) is used to indicate a value relating to the most recent sample, (1) the next older and so on. For the impulse response the suffix (0) means the first sample in the impulse response buffer (i.e. the first that arrived during the analysis process), (1) the next older etc up to (M-1) which represent the most delayed impulse response sample, where M is the number of samples in each impulse response buffer.

Figure 12 shows the flow diagram to calculate each output sample. It comprises a main loop starting at 43 which is executed M times for each output sample by means of the control variable J which is zeroed at 41. The output sample is accumulated into the variable $S_{OUT}$ and so this is zeroed at 42 before entering the loop. The first step in the loop at 43 (for the element J) is to load the impulse response pointer A(J) (being the Jth element of array A). Using this pointer it is possible to load the appropriate impulse response sample from each of the appropriate response arrays. These are referred to as $I_1$ read from A(J)+J (at step 44) and $I_2$ read from A(J)+J+M (at step 45).

The two parts of the input sample $F_1$ and $F_2$ are read from the $F_1$, $F_2$ arrays at offset J at step 46. The two multiply and accumulate steps can be performed to accumulate the output sample into $S_{OUT}$ as shown at step 47. It is then only necessary to increment J (at step 48) and to test this against M (at step 49). When J reaches M the output sample is complete and the loop is finished.

The output sample value may then be fed to the output of the machine (fig 8 items 27 and 28). Input pointers can then be moved on one sample ready for the next input sample.

It should be mentioned that if either of the two simplified processes of fig 2 or fig 4 is to be carried out some simplification of the above processes can be employed. For example, if cross-fading is not to be employed (as described in figure 4) the sample is not divided between $F_1$ and $F_2$ but is simply stored wholly in array $F_1$. Thus all $F_2$ values are considered to be zero and so the memory array is no longer required and any steps relating

to $F_2$ can be by-passed, i.e. step 45, the second read at step 46 and the second multiply accumulate at 47. If the basic linear simulation of fig 2 is required the process is further simplified to eliminate determining the threshold of the input sample, it is simply stored in array F1. Only one address of impulse response is needed so array A is now no longer required and in figure 12 step 43 is not required and the sole base address of the one impulse response is used instead of A(J) at step 44.

It will be appreciated that the number of operations can be substantial as the length of the impulse responses used (M) may typically be 5,000 or longer (although useful results can be obtained with responses as short as for example 50 to 200 steps). Accordingly, and depending on the speed of the DSPs it may be necessary to use more than one DSP to operate in real-time.

Figure 13 shows one possible architecture of a multiple DSP implementation. DSP 51 processes the input sample into the arrays A, $F_1$ and $F_2$ as already described but which are stored in segmented memory arrays 52. This memory is arranged so that it can be wholly accessed by DSP51 for loading with processed input samples, but is separated into sections which can be individually accessed by DSPs 53, 54, 55 etc. Each DSP thus has access to part of each array and for each output sample can perform part of the multiply accumulate loop described in figure 12. The resulting parts of the accumulated output sample are written back to more shared memory 56. DSP 51 (which otherwise is not heavily occupied by the input process) then adds all the separate parts together to produce the whole output sample. Thus for example ten processors (53, 54 etc) could be used so that each performs 500 accumulation steps per output sample, and DSP 51 then has to sum the 10 partial values. Thus 5,000 step impulse responses may be subdivided as appropriate to the speed of the DSP processors. Each DSP 53, 54 etc is effectively executing the same program and so may be fed from either the same or separate program memories 57, 58 etc. It is only necessary to map each part of the memory 52 to appear at the same address location in each associated DSP.

It should be mentioned that there are other ways of dividing up the process which is functionally identical, producing identical output for the same data. For example fig 14 shows a rearrangement of the process so that the bulk of the processing is done for each input sample, accumulating output as the input samples appear. After the Mth input sample is accumulated into the output sample buffer the first output sample is ready for output. Thereafter after each input sample is accumulated, another output sample is available. This arrangement may suit some DSP architectures better depending on the exact nature of the DSP's instruction set.

Methods of generating 3 alternative analysis pulses will now be described by reference to figures 15 to 18. Other methods are clearly possible.

Figure 15 shows details of a digital analysis step tone to be applied to a device under test appropriate to a 16-bit digital audio system. Other bit resolutions would require the amplitude of the steps to be varied appropriately in proportion to the resolution. This figure shows an analysis tone with 128 positive transitions of reducing amplitude designed to obtain 128 impulse responses. It also generates 128 negative going pulses which produce responses which can be ignored, or stored if it is desired to analyse and simulate asymmetric performance.

The digital signal to be fed to the device under test (via a D-A converter if the device is analogue) starts at value zero shown at 100. The maximum positive value the signal can reach is shown at 104 to be value 32,767, and the maximum negative value is shown at 103 at -32,768. These are the limits for a 16-bit linear sampling system. At the commencement of the tone at 101 the signal steps negative to a value of -16,384, and remains at this level for 2n samples. The diagram shows a value of n of 4 but in practice a value of n of 4,000 is typically used. After 2n samples, at 102, the signal steps to +16,384, resulting in a positive step of 32,768 which in magnitude represents the largest amplitude of an individual sample in any 16-bit audio stream. Note that at each transition from negative to positive, the step is always twice the magnitude of the negative value.

After a further n sample, at 105, the value steps to -16256. In fact at each negative going transition (107 etc.) the step is to a negative value 128 less in magnitude than the positive value currently being output. Thus the following negative to positive step (at 106 etc.) is 256 less in magnitude than the previous one.

Thus the sequence of 128 positive steps interleaved between the negative steps have the step amplitudes of 32768, 32512, 32256, 32000, 31744, ... 512, 256.

After the final upward transition to value 128, the final transition at 109 is by -128 to 0. At this point the analysis tone is complete.

The step impulse responses sampled into the analyser may be stored as it arrives (see figure 6) for later processing by the method of figure 7, or the difference signal required may be derived as the data arrives by subtracting the previous sample value from each sample value as it arrives. There are some benefits of postponing the step of deriving the difference signal until later as it is easier to analyse the noise floor of the system with the unprocessed signal during noise removal or level detection operations described later.

Normally the impulse responses derived from the positive going step impulses only will be used, normalised according to the manner described. If the negative going pulses are also to be used to simulate asymmetric devices, the responses resulting from each negative going transition following each positive going one can be stored and normalised by multiplying each sample value by 32768 and dividing it by the (negative) amplitude of the appropriate step transition. Although the negative transitions are slightly smaller than the positive going ones the resulting responses may each be used as if they were for the matching positive impulse transition with negligible loss of accuracy of the simulation.

A further point about the value of n is that this represents the maximum length of impulse response to be derived from the device under test. Although 4000 is a typical

value a larger number must be used if the device under test continues to generate significant response to an impulse for more samples than this. To assist in the later analysis of the tones it is recommended that a multiple of 1,000 samples is used for this value

This signal may be applied directly to a device under test and the resulting impulses recorded for immediate processing and use, or it may be recorded (for example on a digital tape recorder) for application to the device under test at another place or time. In this case the response of the device under test should also be recorded (preferably with the same sample clock as that used for applying the test signal) and may later be fed back into the analyser system described. The analyser can be set to search for the first significant amount of signal which represents the device under test's response to the transition 101, and from this point determine each response to positive transitions spaced at 2n sample intervals. Where the sample clock has differed slightly between the analysis tone and the response sampler, or there is some intrinsic variable delays (for example wow and flutter of a tape recorder) the jitter removal techniques described later can be applied.

The resulting impulse responses are processed by any noise removal algorithms required and the difference signal is derived. The responses are normalised and appropriately windowed for use in simulation.

The process of normalisation requires increasing the amplitude of the impulse responses derived from lower level impulses. It is important not to distort these amplified responses, for example by letting them 'clip' to the peak level storable in the digital representation. A preliminary inspection of the data should be performed to determine any such problem and an attenuation factor generated which is applied equally to all the impulse responses in the set so as to prevent such distortion occurring. This must be done regardless of which method is used to generate the analysis tone.

Figure 16 shows an alternative test signal which can be applied and figure 17

shows a flow diagram describing the process of generating this signal. This method may be used when the device under test is physically connected to the tone generator and analyser device and takes advantage of the fact that the duration of each impulse response can therefore be measured, allowing the impulse response size to be best fitted to the device under test. Where the length of impulse responses desired is limited by other considerations, for example by memory or processing time limitations, it is still possible to wait before applying a subsequent test pulse until the device under test has ceased generating a response to the previous pulse.

Although in this case the sequence is described for a steadily increasing test signal, a decreasing test signal as already described may be used. Values suggested are appropriate to a 16-bit environment where 128 impulses in each direction are required.

Referring to figure 17, at step 71 the initial minimum amplitude value is selected. The output stream from the pulse generator is set to the value $-A_0/2$ at step 72 (typically $A_0$ is 256), producing the output step 81 in fig 16. At step 73 it is necessary to wait for any resulting response from the unit under test to die out. This will be determined by recognising when the non-DC component of the signal has stopped varying or has reached the noise floor which may be determined before the first stimulation is applied. It would generally be advisable to apply to time limit which can be user selectable in case of a varying noise floor causing an indefinite wait.

The test signal is now generated by stepping the output stream by the amplitude A, by stepping in a direction to cross the zero value, as described at step 74. This is shown at 82 in fig 16 for this first value. The resulting output of the device under test is now monitored and stored as the step impulse response (step 75).

At step 76, value A is tested to see if it has reached the maximum step desired (typically 32,768) and if not it is increased (typically by 128) to the next amplitude to test (step 77). The process then loops back to step 73 where any residual response to the

stimulation is allowed to die out, then the output is stepped again, this time in the opposite direction. This is shown at 83 in fig 16. Once again the output stream is monitored and the signal is stored.

As for the previous signal of figure 15, the resulting impulse responses are processed by any noise removal algorithms required and the difference signal is derived. The responses are normalised and appropriately windowed for use in simulation.

Although step impulses are normally used, it is possible to apply simple impulses as suggested in figure 3, and figure 18 shows how the algorithm of figure 17 may be modified to carry out this process. Once again this shows the application of steadily increasing pulses but of course the larger impulses may be applied first. There is a slight benefit of steadily increasing the pulses as any residual effect from a previous impulse will be slightly less than if the pulses are decreasing in value but in practice there are benefits in level setting for example which outweigh this.

In figure 18 at step 61 the initial amplitude is typically set to 256. At step 62 it is desirable to wait for any residual effect of a previous signal passing through the unit under test, as some effect devices may continue to generate output for some time after stimulation, for example due to resonances or reverberations. This process is done by monitoring the return signal from the device under test and observing the noise floor. If this is decaying over a short space of time the process simply waits for the noise floor to become stable.

At step 63 a test pulse of the desired amplitude is emitted by setting the output stream to the value A in one sample period and back to zero at the following sample. At step 64 the returning stream is monitored and stored (usually into RAM) until the time limit set by the implementation is reached. This is determined by the number of steps which the simulator can process in real time, or can be limited by memory available or be further limited by user intervention to minimise processing requirements. It should also be

noted that the process of step 62 can also be followed to determine when there is no significant further response and further used to shorten the sampling process.

Once the sampling is complete the amplitude is tested at 65 to determine if the process is complete (usually when the signal has reached 32678. If not, the next higher level of amplitude can be loaded into A (typically increasing it by 256) and the loop repeated. Note that an impulse of 32768 cannot in fact be generated in a 16-bit system but the maximum value 32767 can be used with insignificant loss of accuracy.

A useful refinement to any of the above analysis pulses is to allow the system to generate a continuous stream of pulses at user definable amplitudes solely for the purpose of allowing the operator to select the optimum levels of signal to pass through the device under test.

It should be noted that the step of waiting for any residual stimulation of the device under test (shown at step 62 of fig 18 and step 73 of fig 17) may be replaced by a fixed wait period in many instances where there is not significant energy storage in the device under test. This has a benefit that the output test signal becomes the same for any test and as in the case of the signal of figure 15 may be recorded (preferably in digital format) for application to a device remote from the analysis machine. The resulting output from the device under test may also be recorded and can later be analysed by the analysis process. The only significant alteration to the process of analysis is that instead of generating the test pulse it is necessary simply to wait for any significant response to appear in the recorded stream and store this and the following impulse response as being the response to the first test pulse, then similarly wait for the appearance of further responses to later pulses and thus obtain a complete set of impulse responses. This is useful as an operator may simply carry a tape of the test stream and if he encounters a device which he wishes to analyse he simply plays the tape through the device and records the result for later analysis and simulation. A useful refinement to improve this process is to precede the test signal with a short burst of tone which both can be used for level setting and can be

recognised at the analysis stage and taken as a trigger to start the process of looking for response signals at a known period after the tone burst.

Although the sampled effect is shown as an analogue device, a digital processor may be sampled by applying the sample impulse directly to the digital input and sampling directly the output impulse response.

**Improving noise**

A potential problem with the system is that significant noise generated by the device under test will appear as noise in the simulated effect. This can be made worse when using impulse responses derived at low levels of test. However since many effects become linear as the level through the device decreases it is often just necessary to use a set of impulse responses derived at relatively high levels, and below this threshold of linearity, to use the impulse response derived at the highest linear level in place of all lower impulse responses. This can be done under manual intervention from the operator who can choose a balance between desirable non-linearity and acceptable noise by auditioning the effect of selective replacement.

Where it is not possible to achieve a desirable balance because it is desired to preserve lower level non-linearities where noise is a problem, it is possible to selectively modify parts of the impulse responses derived at low levels by replacement with matching parts of the responses from higher level impulse responses, where the areas to be replaced are determined by evaluating the absolute amplitude of each section of the response and replacing it where the impulse response is seen to be near the noise floor.

Figure 19 shows some details of this process. A higher level impulse response is shown at (a) and a lower level one at (b). An envelope 91 (at (c)) is generated representing the average level of a local region of the impulse response (b). This is evaluated by calculating the RMS value of the nearby samples, weighted towards the current time for each point in the envelope. In practice this may encompass 400 to 500 samples if the

impulses used are say 5000 samples long, or smaller ranges if shorter impulses are to be used. The envelope 91 is compared with a threshold 92 which may be user determined or estimated by comparing with the noise floor determined either by monitoring the device output under no signal conditions. In fig 19 (c) it can be seen that the example envelope 91 falls below the threshold at 93 and rises again above it at 94. From this a 'cross-fade envelope is generated (d). Using this envelope the impulse response (b) is selectively replaced with impulse response (a) with a soft crossfade of several milliseconds at each end of the replacement (shown as the ramps 95 of the cross-fade envelope) to generate a new impulse response (e) where the lower level area is replaced by the lower noise floor impulse response taken at the higher level (a).

The new impulse response is generated according to the formula

$$r=e.a + (1-e).b$$

where e is the cross-fade envelope value, a is the sample value from the higher level impulse response and b is the sample value from the lower level impulse, and r is the resultant sample to replace in the lower level sample. The period (.) represents multiplication. This provides a cross-fade to the higher level impulse response where the lower level signal was below the threshold.

To determine the noise floor automatically it will be seen that for the impulse responses taken at lower levels there will be a level which the envelope never drops below due to noise. The threshold can thus be set say 50% above this and applied progressively from a higher level sample down to the lowest level. It is appropriate to start the process at the impulse response some 12dB below the maximum, in other words that sampled with a sample pulse about a quarter of the amplitude of the highest sample impulse used.

## Length of impulse responses and processing power

The impulse response lengths required depend on the energy storage characteristics

of the effect sampled. Typically an equaliser, valve amplifier or speaker/microphone combinations in short reverberation environments can be simulated with impulse times of up to 1/10th second, or for example 5,000 samples. Each output sample will require the accumulation of 5,000 values of input sample multiplied with 5,000 impulse response samples, or 250 million operations per second assuming a 50,000 sample per second sampling rate. Thus the simple case of fig 2 requires 250 million multiply accumulates (MAC) operations on linear arrays of data, while the process shown in fig 12 requires correspondingly more steps to be repeated this many times.

Some valve processors and tape-recorders have very short impulse responses and a useful simulation can be achieved with responses as short as 200 samples.

To simulate fully reverberant effects, impulse responses of several seconds can be needed resulting in a proportional increase in processing power. This is quite possible within a network of DSP chips. To make the best use of a particular hardware implementation however the simulator should be arranged to switch amongst the three simulation methods described: the linear simulation of fig 2, the simple non-linear simulation of fig 4 and the interpolated simulation of fig 5 (shown in greater detail in figs 8 onwards). This means that in simulations where non-linearity is not required more processing power is available for longer impulse responses and therefore longer reverberant periods.

## Windowing of impulse responses

It should be noted that where an effect is sampled but the impulse response exceeds the length of sample which it is possible to calculate in real-time in a particular hardware implementation, it is necessary to truncate the impulse response by windowing the response, i.e. effectively fading off the last 1/20th second or so linearly to zero. In fact all sampled impulse responses should be windowed in this way to prevent any glitch effects from suddenly truncated noise signals. Where impulse lengths are short the fade out typically would be across the final quarter of the response signal.

It is also beneficial to apply a fade-in ramp over the first few (for example, 10) samples of the derived impulse response, and for this purpose it is desirable to store a few samples before the actual impulse response is received so this fade-in takes place over the residual noise of the system.

## Editing impulse responses

### Trimming the start and end

There is always some delay between the application of an impulse to a device and the output response. This results in an equal delay in the simulation. Sometimes the effect can be improved by removing or reducing this delay and in any event this shortens the sample to reduce computational requirement. It is simple to arrange for the operator to trim off samples from the front of the sample - the effect of which he can audition to his taste, or a threshold level can be set on a response to automatically trim off any initial response below this 'noise' threshold. This threshold would typically be applied to the impulse response derived from the highest level sampled signal and once determined, the same amount is trimmed off the start of the whole set of impulse responses.

### Frequency shifting

Interesting variations of the sampled effect may be made by re-sampling each impulse response to a higher or lower frequency using standard re-sampling algorithms. The effect of each change can be auditioned to the taste of the operator. This allows various effects, such as for example the resonances in the sampled effect being matched to dominant frequencies in the signals to be processed.

### Combination of effects

It is possible to simulate the effect of passing a signal through two successive effects by taking each impulse response of the first effect and passing it through the simulation of the second effect to generate a new impulse response for that sample amplitude. This is done for each impulse response of the first effect to achieve the same

number of new impulse responses representing the combined effect. In the case of the simple method of fig 2 this represents s simple convolution of the impulse responses.

**Interpolation and extrapolation effects**

The set of impulse responses representing the range of levels passing through an effect embodies the non-linear characteristic of the sampled effect. New and interesting effects can be achieved by partially linearising the effect. To do this a subset representing a range of the original set is taken and a new complete set of impulse responses is generated by interpolation of each sample step through the set of impulse responses.

It is also possible to make the non-linearity more extreme by extrapolating beyond the original range. This can result on extreme values of samples and generally the whole sample set will have to be attenuated to keep the output within acceptable limits.

After any such recalculation the operator can again audition the effect to achieve a desired effect. The extrapolation effects will generally become very strange but small amounts of extrapolation may generate desirable distortions.

## Arithmetic

As with all good signal processing practise care must be taken with rounding or truncation of digital value. It is best to preserve precision of all calculations to, for example, 32-bits if fixed point arithmetic is used of 24-bits of mantissa if floating point is used. Final digital output can be reduced to the desired digital output format using appropriate and known bit reduction techniques.

## Storing only the first n responses of a set

It has been stated that at low levels the impulse responses can become lost in the noise of the device under test. Accordingly the operator can determine the lowest level impulse response which it is desired to use. Below this in the simulation, the lowest specified impulse response is used for all lower sample values.

Accordingly it is not necessary to store the data for the impulse responses that will not be used, but simply to store an indication that the last specified response be used for all lower level samples.

When reloaded for implementing a simulation according to the embodiment of the invention described, the impulse response derived from the lowest level exciting pulse stored is simply replicated to complete the set.

It should be mentioned that an alternative embodiment may change the simulation algorithm so that although sample levels above the lowest level impulse response are subject to selection and interpolation between the appropriate higher level impulse responses, those below the lowest level impulse response present are simply applied to this lowest level response without the need for interpolation. In this situation there is no need to replicate the lowest level response defined in the stored set of data.

## Precision of Sampling Clock

In generating a set of impulse responses corresponding to different amplitude impulses it is important that each impulse response is closely correlated with the others. In other words the relationship between the exciting pulse and the resulting response of the device under test must be strictly linked. This requires that the digital input sampling system is locked to the digital output system generating the analysis tone. In addition long term clock accuracy should adhere to good audio design practice so that some time into each impulse response, impulse samples remains correlated between different impulse responses in the set.

In the event that this requirement cannot be met it is still possible to extract a usable impulse response set by means of jitter removal.

## Jitter Removal

Where it is impossible to guarantee high accuracy between the timing of the

analysis tone and the resulting impulse responses, for example where the impulse response is recorded and reproduced later for analysis, or where the device under test introduces small timing errors (for example when sampling an analogue tape recorder with its intrinsic delay between recording and replaying), it is necessary to re-correlate the impulse responses.

Figure 20 shows two successive impulse responses at (a) and (b). These are samples at the time intervals indicated by the vertical lines, giving digital samples as shown at (c) and (d). Although the impulse responses are apparently very similar (as would be expected where the exciting impulse was only slightly different in amplitude), due to the fact that time correlation has been lost between the analysis tone and the resulting impulse response it can be seen that the digital signals appear very different although they represent a broadly similar underlying (analogue or implicit) signal. They are thus not suitable for use as members of an impulse response set until the timing inaccuracy is removed.

This may be corrected by up-sampling the digital signal (to n times the original rate, where the diagram shows the case for n=3) by known means (typically accumulating a 'sinc' function for each sample in the digital stream) to achieve the digital signal as shown at (e) and (f), where the interpolated new samples are shown as thinner vertical lines and the underlying (implicit) wave-form is shown dotted.

It is now possible to look for a recognisable characteristic of each signal and typically this can be done by looking first for the peak amplitude of the first impulse response. This is clearly the sample shown at 110. This impulse response can now be decimated to the original sampling rate simply by taking every *n*th sample to generate the new digital signals at (g) and (h).

For each subsequent impulse response it is now possible to look for the largest amplitude sample with a matching sign to that of the first impulse response (for example

that shown at 111), and similarly decimating each impulse response so that this highest point is now precisely correlated with that of the first.

In fact, up-sampling by 64 times together with this pattern matching algorithm gives good results for the example of analysing an analogue tape-recorder, where the impulse responses have a clear initial peak. Higher up-sampling rates may be used if higher precision is desired. Other pattern matching algorithms may be used including allowing the system operator to match the patterns by hand and eye by overlaying images of the digital representations on a display screen. This would be more appropriate for extreme devices under test with very complex impulse responses.

## Smoothing over a range of impulse responses in the set

An impulse response measured at one time may vary slightly from one taken at another time due to various random variations in the device under test. For example when analysing an analogue tape recorder instantaneous gain can vary due to inconsistencies in the tape medium.

Ideally a number of measurements should be taken and the impulse responses for each amplitude excitation pulse can be simply averaged on a sample by sample basis to smooth out these variations. This also reduces the effects of noise in the device under test. For the example of an analogue tape recorder typically 16 sets of measurement may be taken but this depends on the device and auditioning the results obtained. The number of measurements can be increased until a desirable quality of simulation is obtained.

A faster and more convenient way of achieving almost as good results can be achieved by recognising that each impulse response of the set obtained in a single analysis run with the analysis tone differs only slightly from other responses near to it in the set. This is because the variation in impulse response encapsulating the non-linear characteristic of the device under test is generally a gradual one.

Accordingly it is possible to average a number of adjacent impulse responses (on a sample by sample basis) to create a new impulse response. Typically, for the example where there are 128 impulse responses in the set, and it is chosen to smooth over 8 impulse responses: The first impulse response is replaced by the average on the first 8 impulse responses. Then the second is replace by the average of second to the ninth response, the 3rd by the average of the 3rd to the 10th, etc, until the 120th is replace by the average of the 120th to the 128th. This final average is also used to replace response 121 to 128, resulting in a linear lower end of the simulation.

Where the lower level impulse responses are not available because they have not been kept, (for example if they were not stored after the operator decided that they were too near the noise of the device under test), the averaging process must stop n responses from the end, where we are smoothing over n responses. The set of responses is thus reduced by (n-1) and the new last response is used for all lower level samples in the simulation.

## Selecting Between Impulse Responses based on Envelope

The non-linear synthesis has been described where the selection between impulse responses of a set and the relevant interpolation is based on the instantaneous sample value for each sample, as shown in figure 10.

Useful variations in the simulated effect can be achieved by substituting for the sample level the envelope of the audio signal being processed. This can be implemented by providing user control over two additional parameters, referred to here as 'attack' and 'decay'. The effect already described in fig 10 (for the system where only the amplitude of the input sample is considered for the selection criterion) is produced when both these parameters are set equal to 1.

The envelope may be generated by maintaining an ongoing variable named here 'env'. At the start of the process this may be initialised to zero and will quickly attain its

correct value.

The flow chart for calculating a new value for the envelope 'env' for each input sample is shown as figure 21.

For each sample the sign is removed at step 121 by taking the absolute value of the sample and assigning it to 'v'. The existing envelope is then allowed to decay at step 122 according to the value of the 'decay' parameter. This is an exponential decay towards zero. If 'v' does not exceed the decayed envelope value 'env' we have the value to be used. If it does exceed 'env', determined at step 123, then the new value for env is calculated at step 124. Effectively the value of env is increased towards the value v according to the 'attack' parameter. This would represent an asymptotic growth if the incoming sample values were consistently higher than 'env'.

Finally at 125 the value of env is used instead of the sample value in the algorithm of figure 10 to determine the impulse response to be used in the system and the proportions k (in fig 10) of each adjacent impulse response to use. Finally the values S.(1-k) and S.k are calculated at the penultimate step of figure 10, using the input sample value for S as before to generate the $F_1$ and $F_2$ values used for calculating the output sample value. The system is ready for the next input sample.

A useful improvement is to store the previous n input samples, and after calculation of the 'env' variable based on the current input sample, to use the input sample value read in n steps previously to generate the output sample, saving the current input sample for n iterations. This allows a sudden increase in input signal to allow the 'env' variable to increase appropriately over a number of steps before the first high level input sample is actually applied to the simulation algorithm. A disadvantage is that this introduces an overall delay into the system. Once again this value of n is usefully made a user controllable value.

Typical values of attack and decay are 10 and 1000 respectively resulting in a rapid adoption of a higher level impulse response when the input signal increases in general amplitude coupled with a slower return to lower amplitude values when the general signal level decays. 'n' can be made variable from 0 up to several time 'attack'.

Where the original device under test contained in-built audio dynamic compression characteristics (e.g. a 'compressor limiter' device) this approach of impulse response selection based on envelope more accurately simulates the way the device under test alters its tonal characteristics and gain at different levels of applied signal.

## Order of Processing Derived Impulse Responses

There has been described a number of operations to be performed on the impulse response data resulting from analysing a device under test. It is necessary to first apply the de-jitter algorithm if this is necessary. Noise measurement and substitution to reduce noise is best performed next. Since the signal has not yet been normalised it is necessary to reduce a higher level impulse response data in proportion when substituting in a lower level impulse response. At this stage the difference signal should be derived if a step analysis pulse was used. Following this the impulse responses should be normalised, and then any smoothing between responses is performed. Finally the responses should be windowed as described.

## Further Uses

The process described can be used to simulate effect which are asymmetric by also taking into account the sign of the signal to be processed and taking separate analysis samples for positive going test pulses and negative going test pulses. This asymmetric processing could be appropriate, for example, to simulation of high sound pressure level effects in air where the sound carrying capacity of air is asymmetric.

A further use of the process of selecting between impulse responses is for using some other characteristic than the amplitude of the incoming sample to control selection.

For example a number of different effects can be placed into each impulse response memory and be selected between (including using the cross-fading technique) under user control or in a repetitive manner using a control oscillator. In this way a time varying effect can be simulated, for example a rotating Leslie loudspeaker cabinet or a varying flanger or phaser effect. The required impulse responses can either be calculated to generate an effect or an existing unit can be sampled at a number of different settings representing a range which the effect is normally used to sweep through. Thus a Leslie loudspeaker can be analysed at a number of different static positions of the rotating speaker and the resulting set of impulse responses stored. Then cycling through the responses will simulate rotation of the speaker (including the doppler effects of the moving speaker as different impulses responses will have different delays built in representing different direct and indirect signal paths from the loudspeaker analysed).

A refinement of the process allows the combination of non-linear effects with time varying or user controlled effects. In this case instead of one set of impulse responses which are amplitude dependent, a number of sets are stored. The amplitude of the incoming signal determines which impulse response of a set to use, and the time varying or user adjusted parameter selects between sets. To perform smooth cross-fading between effects the interpolator function of figure 5 is enhanced to provide a two-dimensional, or bilinear, interpolation between 4 impulse responses with one dimension dependent on signal amplitude and the other dependent on the other parameter. It is of course possible to increase the number of parameters which can be varied simultaneously still further. This is limited by the processing power required to perform the multi-linear interpolation and the storage capacity for the number of impulse response sets required.

Although a monophonic system is described typically two units will run in parallel to allow stereo in and stereo out. Often the input signal will be the same applied to both channels to generate stereo simulated effects from monophonic sources.

It was mentioned that audio processing is used in film dubbing. An example of use

of the invention is as follows: Once the effect on an actor's voice has been decided upon in a film production to match the studio recording to the appropriate sound for the scene, the entire process through which the voice track is passed can be analysed and stored. In this case whenever it is necessary to re-record the sound track, for example when dubbing into a foreign language, the effect can be recalled and applied to the relevant speech for the appropriate scene. Thus a film would be made available for dubbing with the audio process for each scene and each voice stored and indexed to speed up the dubbing process.

## Non real time and general purpose computers

Note that it is also possible to process in non-real time using less hardware and this can be done on typical general purpose desk-top computers. However the best use is achieved when operating in real-time whether this is on a high performance general purpose computer implementing the algorithms described or by means of dedicated multiple DSP architectures.

## Deriving impulse responses from virtual systems

It should be noted that as well as sampling existing effects it is quite possible to generate a computer model of a new device and calculate a set of impulse responses. These may then be loaded into the simulator to allow the effect to be auditioned in real-time. In this way the simulator can emulate arbitrary digital effects such as equalisers, or simulated physical models e.g. room simulations, and especially non-linear devices such as amplifier or loudspeaker simulations.

In the case of simple equalisers which are linear in character only one impulse response is generated for any chosen equaliser. These can be calculated and loaded rapidly to allow real-time variation of equaliser characteristics. The simulator thus provides a powerful simulator of a wide range of equaliser devices complete with real-time user control of parameters. In practice when a parameter is varied the new impulse response is calculated and loaded and a cross-fade can be performed to the new effect to remove switching effects when parameters are varied. This can be extended to include non-linear

processes by using the multi-dimensional approach described.

# CLAIMS

1. A method of simulating an audio effect processor comprising the steps of:

    a) storing the impulse response of the audio processor for at least two impulses;

    b) repeatedly assessing a characteristic of an input signal;

    c) selecting at least one of the impulse responses to apply to the input signal in dependence on the result of the assessment; and

    d) applying the selected impulse response to the input signal to derive an output signal.

2. A method according to claim 1, wherein the step of storing the impulse response comprises storing at least two sets of digital samples representing the at least two impulse responses and the step of applying the stored impulse response comprises the step of convolving each of a first set of digital samples representing the assessment of the characteristic of the input signal with the selected set of digital samples representing the selected impulse response appropriate to the characteristic to give a second series of digital samples representing the output signal.

3. A method according to claim 1 or 2 in which the step of assessing a characteristic of the input signal comprises assessing its amplitude.

4. - A method according to claim 3, in which the step of selecting an impulse response to apply to the input signal comprises determining whether the amplitude of the input signal is above or below a predetermined threshold.

5. A method according to claim 3, in which the step of selecting an impulse response to apply to the input signal comprises determining whether or not the amplitude of the input signal falls within a predetermined range, applying more than one impulse response to the input signal if the result of the determination is that the amplitude of the input signal

does fall within the predetermined range and deriving the output signal therefrom.

6. A method according to claim 5, in which the more than one impulse responses applied to the input signal are applied in proportions which sum substantially to 1.

7. A method according to claim 6, in which the proportions of the impulse responses applied to the input signal are dependent on the position of the amplitude of the input signal within the predetermined range.

8. A method according to claim 1 or 2 in which the step of selecting an impulse response to apply to the input signal comprises the step of detecting a user input and selecting the impulse response in dependence thereon.

9. A method according to claim 1 or 2 in which the step of selecting an impulse response to apply to the input signal comprises the step of monitoring a time dependent variable and selecting an impulse response in dependence thereon.

10. Apparatus for simulating an audio effect processor comprising:

(a) means for storing the impulse response of the audio processor for at least two impulses;

(b) means for repeatedly assessing a characteristic of an input signal;

(c) means for selecting at least one of the impulse responses to apply to the input signal in dependence on the result of the assessment; and

(d) means for applying a selected impulse response to the input signal to derive an output signal.

11. Apparatus according to claim 10 wherein the means for storing the impulse response comprises means for storing at least two sets of digital samples representing the respective impulse responses and the means for applying the stored impulse response comprises means for convolving each of a first series of digital samples representing the

assessment of the characteristic of the input signal with the selected set of digital samples representing the selected impulse response to give a second series of digital samples representing the output signal.

12. Apparatus according to claim 11 or 12 in which the means for assessing the characteristic of the input signal comprises means for assessing the amplitude of the input signal.

13. Apparatus according to claim 12 in which the means for selecting an impulse response to apply to the input signal comprises means for determining whether the amplitude of the input signal is above or below a predetermined threshold.

14. Apparatus according to claim 12 in which the means for selecting an impulse response to apply to the input signal comprises means for determining whether or not the amplitude of the input signal falls within a predetermined range and the means for applying more than one impulse response to the input signal does this if the result of the determination is that the amplitude of the input signal is within the predetermined range.

15. A method according to claim 14 in which the more than one impulse responses applied to the input signal are applied in proportions which sums substantially to 1.

16. Apparatus according to claim 15 in which the proportions of the impulse responses applied to the input signal are dependent on the position of the amplitude of the input signal within the predetermined range.

17. Apparatus according to claim 11 or 12 in which the means for selecting an impulse response to apply to the input signal comprises means for detecting a user input to which the selecting means is responsive.

18. Apparatus according to claim 11 or 12 in which the means for selecting an impulse

response to apply to the input signal comprises means for monitoring a time dependent variable to which the selecting means is responsive.

19. A method according to claim 1 comprising the step of storing the impulse response for a plurality of different audio processors.

20. Apparatus according to claim 10 comprising means for storing the impulse responses for a plurality of different audio processors.

21. A method for storing the impulse response of an audio processor for use in the method of any of claims 1-9.

22. Apparatus for storing the impulse response of an audio processor for use in the apparatus of any of claims 10-18.

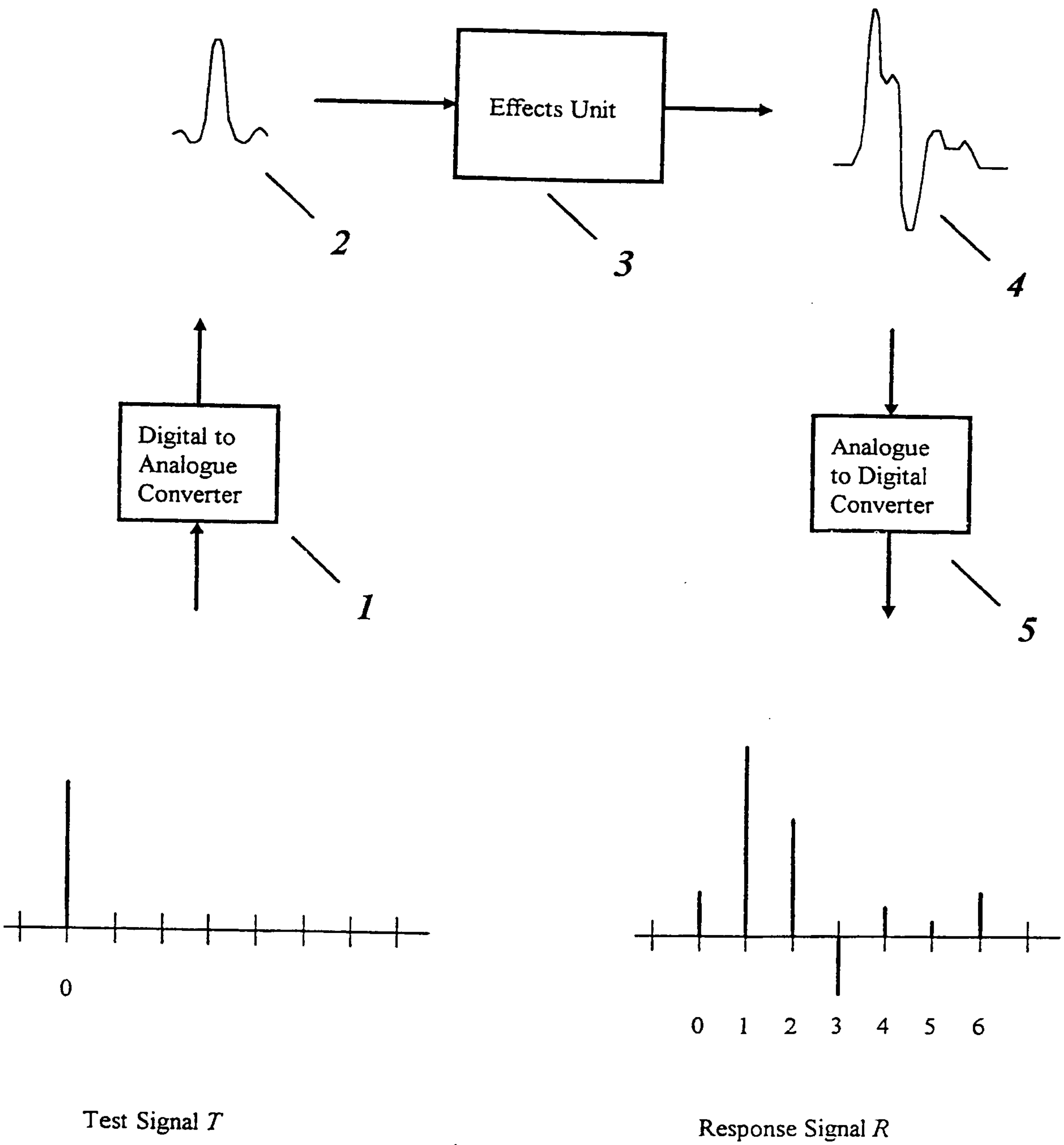

Effects Unit
2
3
4
Digital to Analogue Converter
Analogue to Digital Converter
1
5
0
0 1 2 3 4 5 6
Test Signal $T$
Response Signal $R$

Fig 1

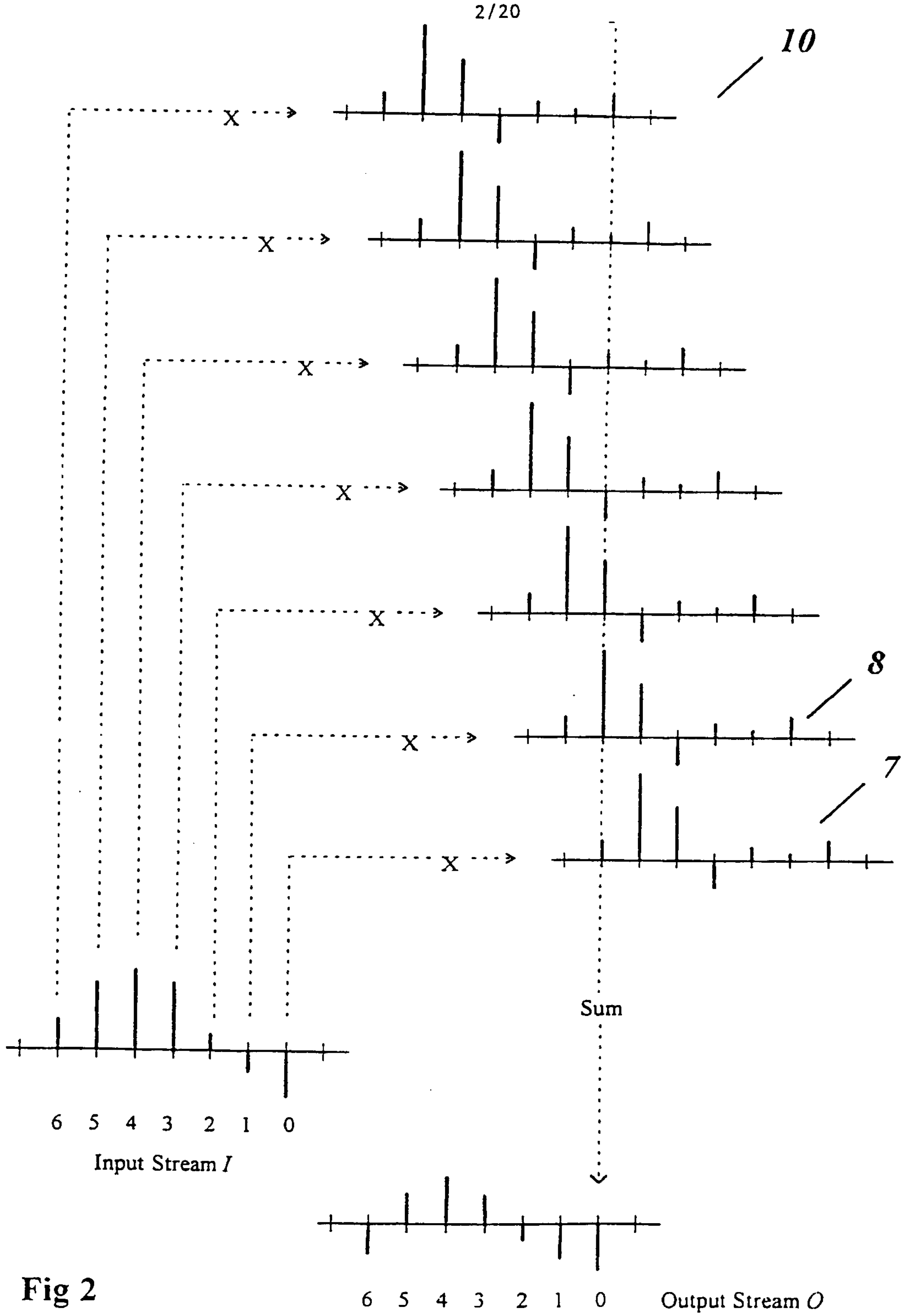



Fig 2

Fig 3a

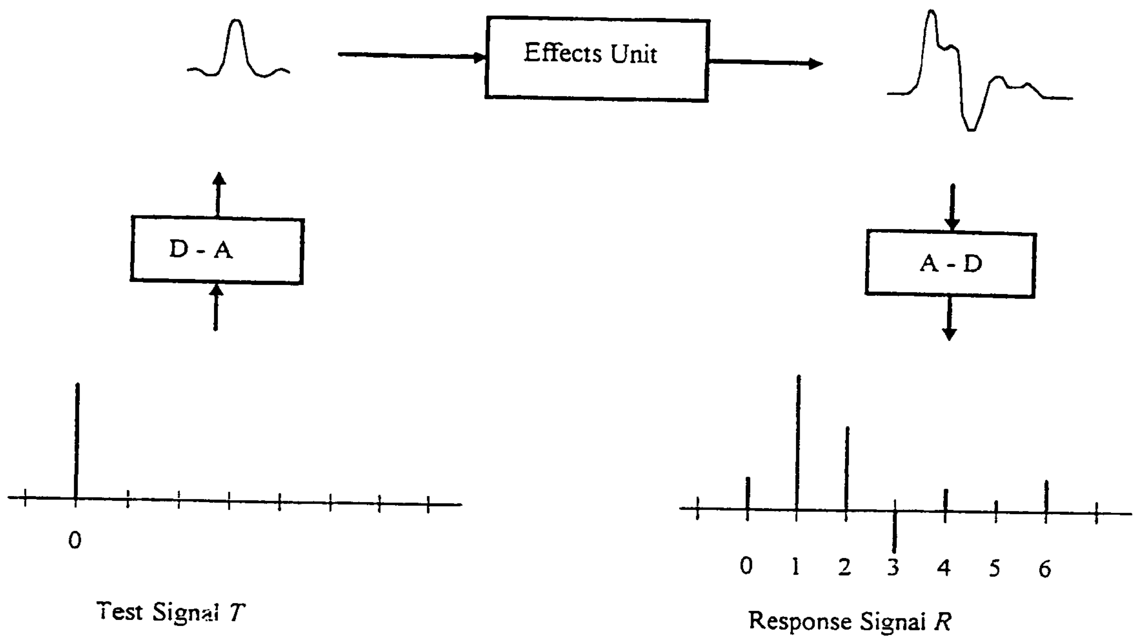
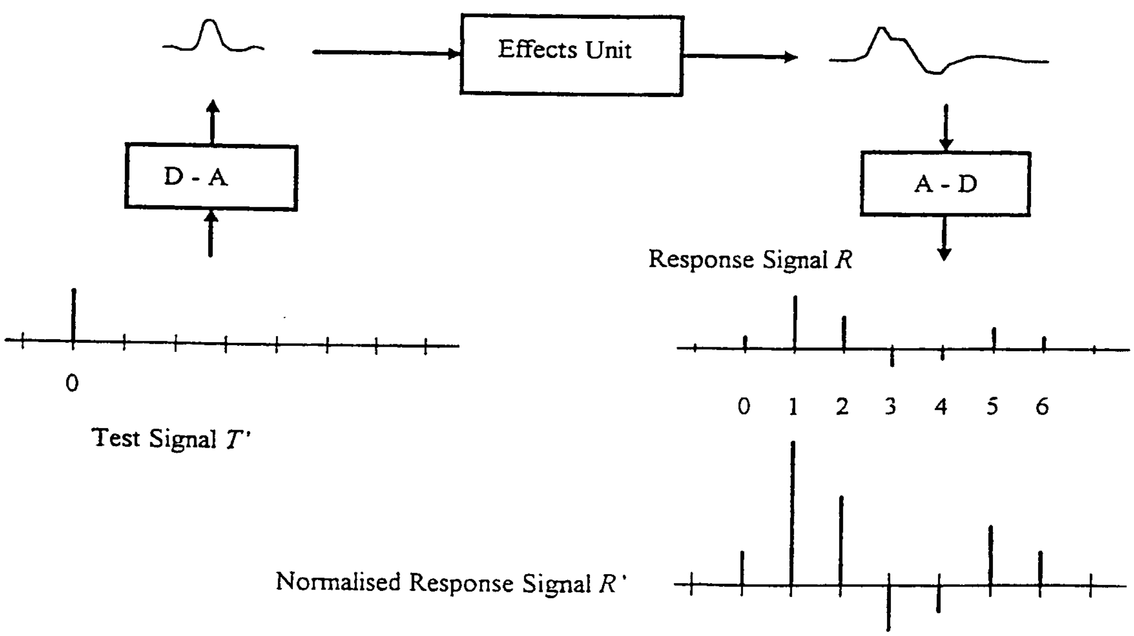

Effects Unit
D - A
A - D
0
0 1 2 3 4 5 6
Test Signal $T$
Response Signal $R$

Fig 3b


Effects Unit
D - A
A - D
Response Signal $R$
0
0 1 2 3 4 5 6
Test Signal $T'$
Normalised Response Signal $R'$

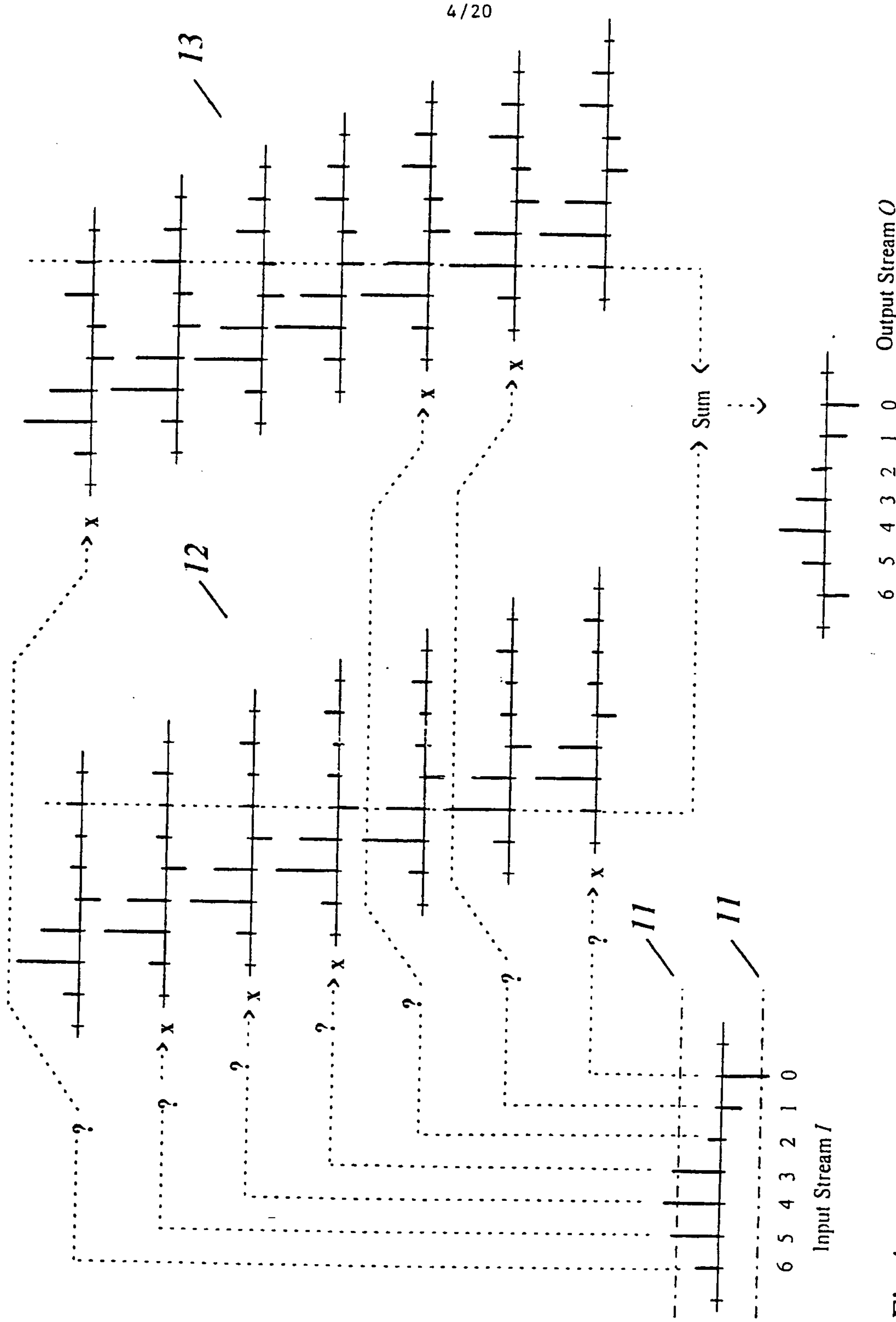

13
12
11
11
Sum
Output Stream O
6 5 4 3 2 1 0
Input Stream I
6 5 4 3 2 1 0

Fig 4

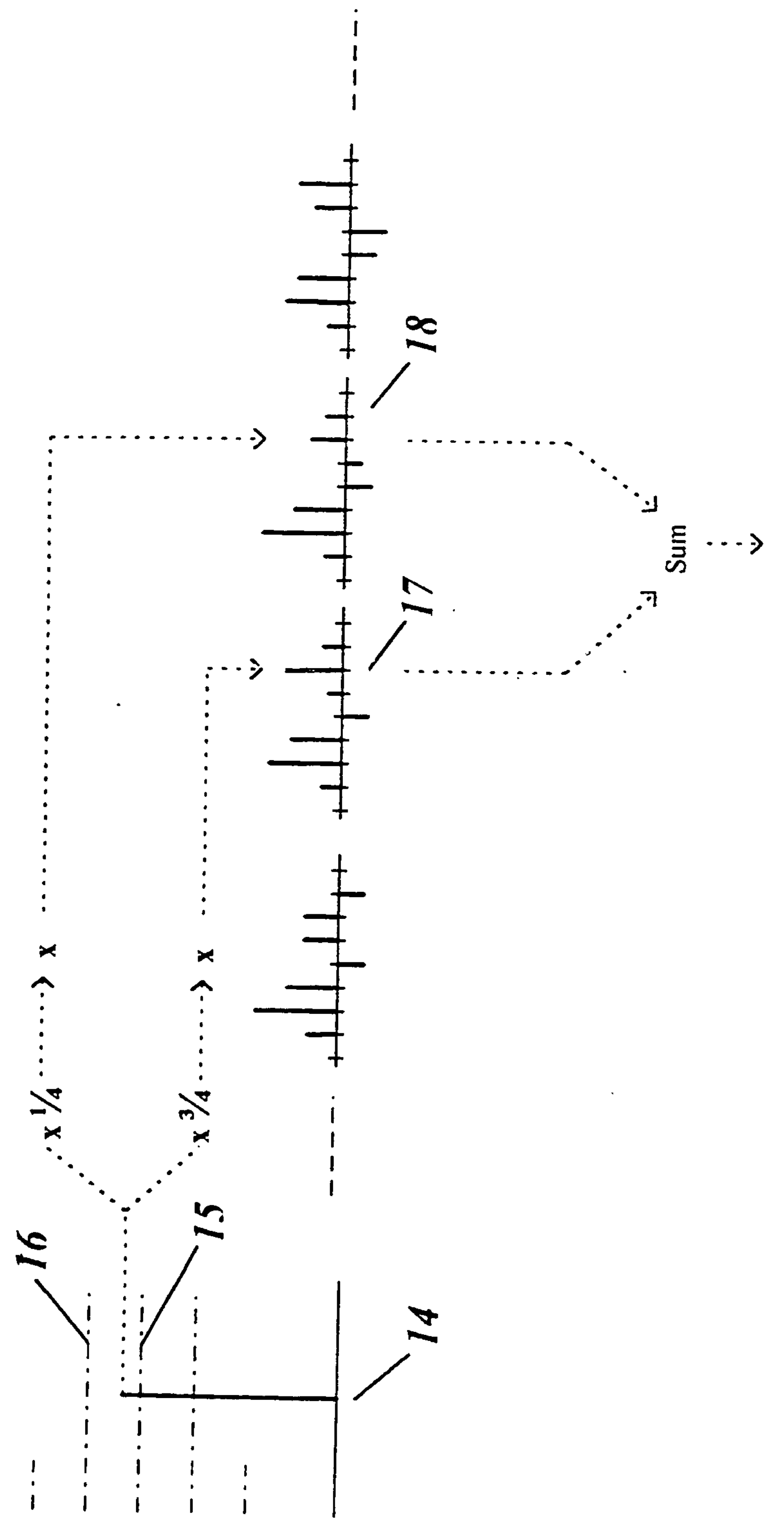
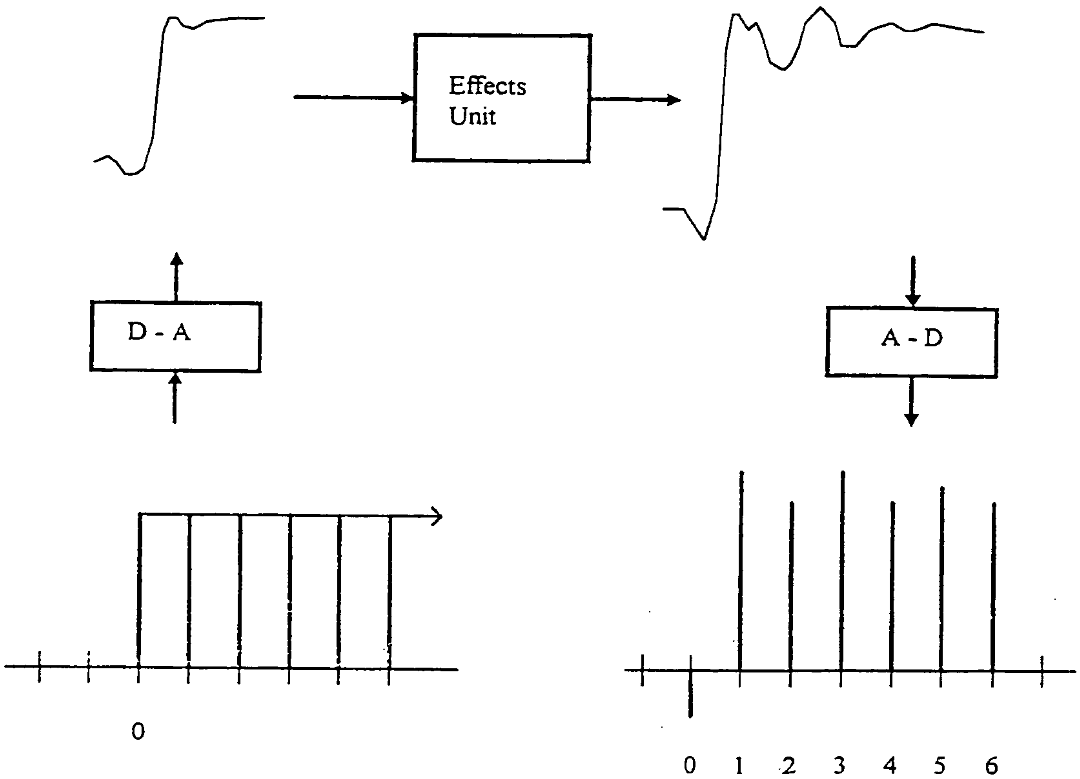



Fig 5

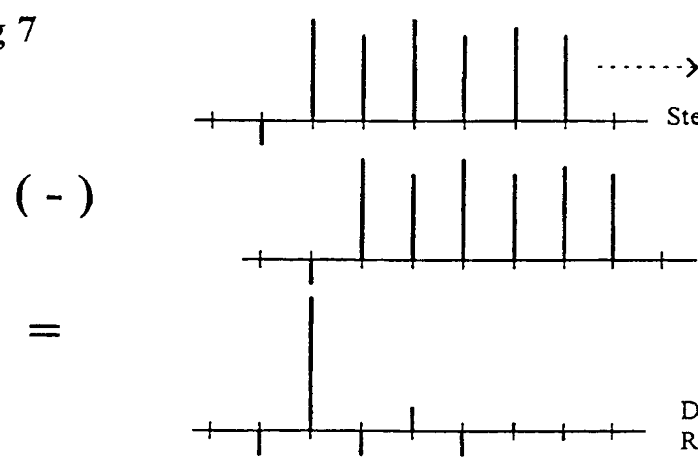
Fig 6




Test Signal *ST*

Step Response Signal *SR*

Fig 7




Step Response Signal *SR*

Shifted Step Response Signal *SR'*

Derived Impulse Response Signal *R*

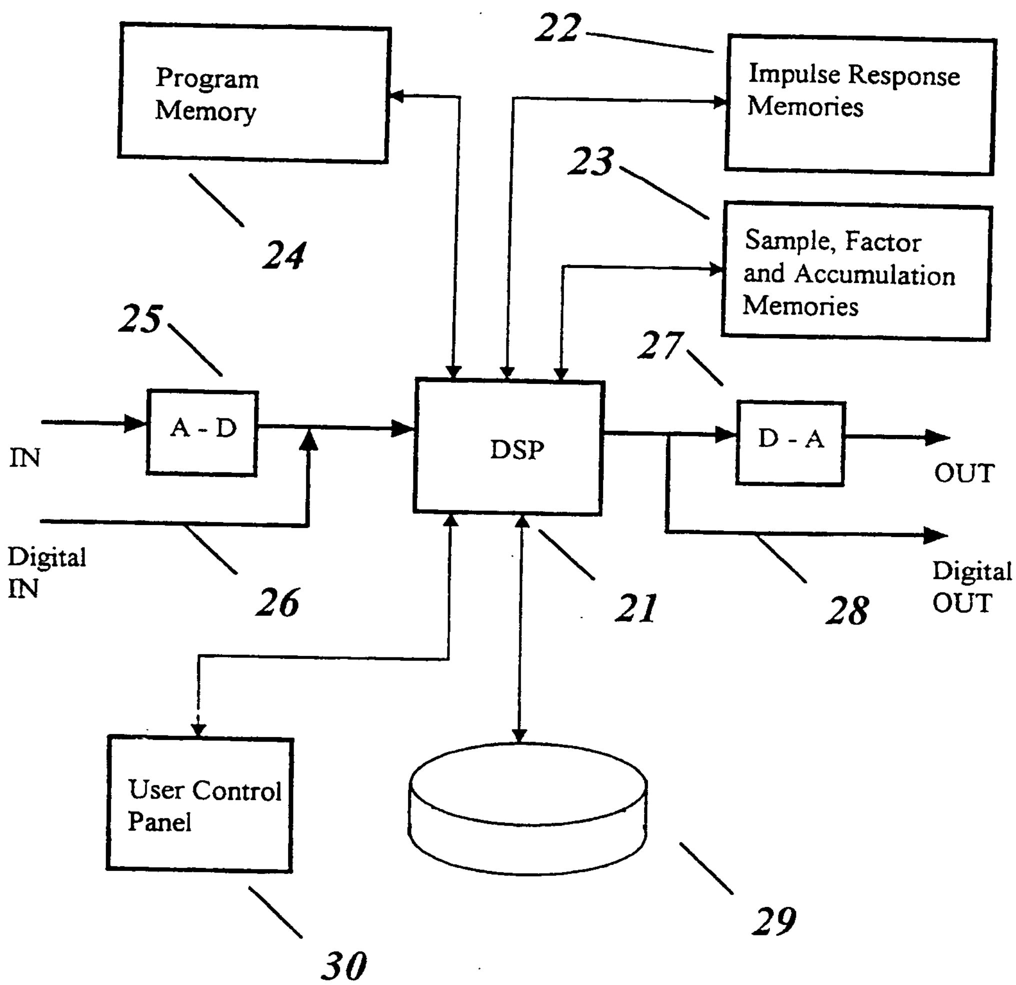

22
Impulse Response Memories
Program Memory
23
Sample, Factor and Accumulation Memories
24
25
27
A - D
DSP
D - A
IN
OUT
Digital IN
26
21
28
Digital OUT
User Control Panel
29
30

Fig 8

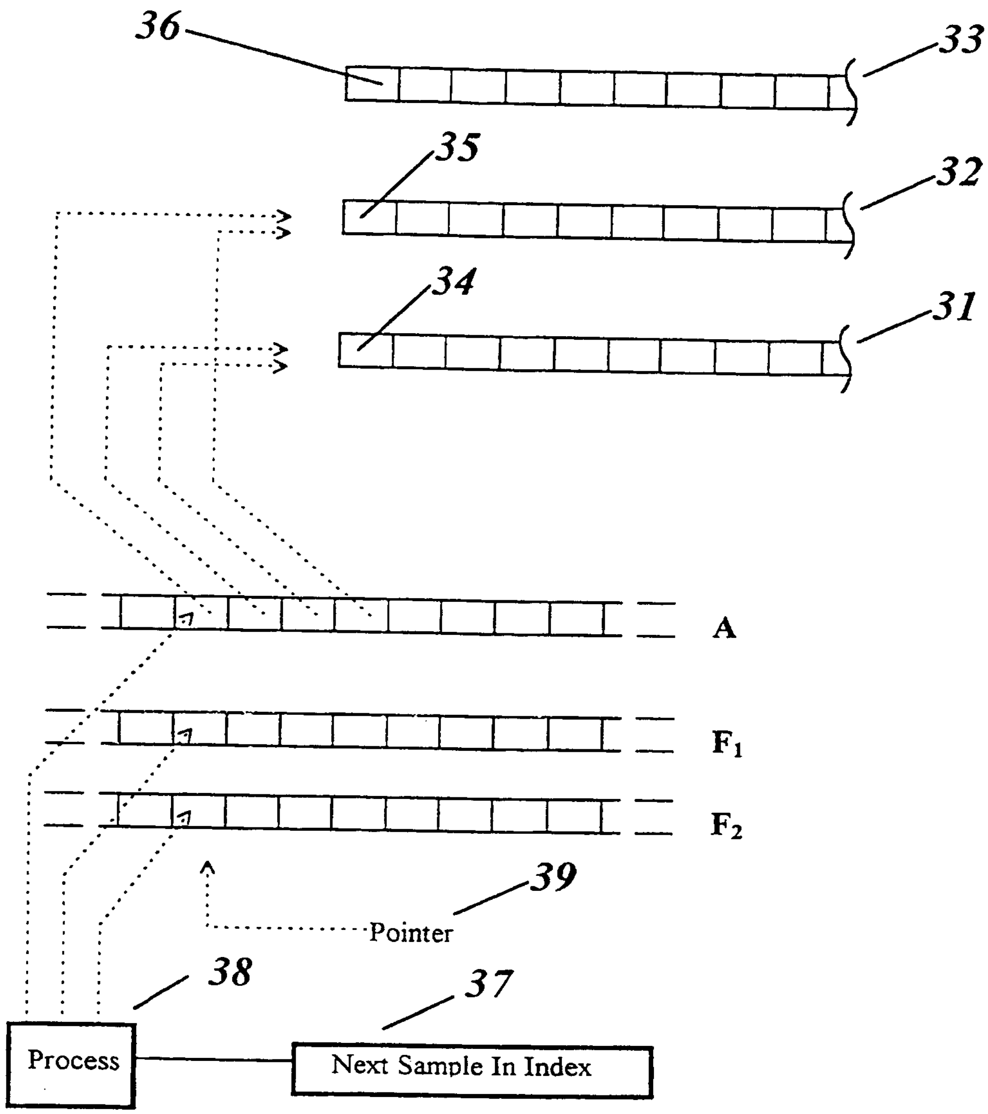

36
33
35
32
34
31
A
$F_1$
$F_2$
39
Pointer
38
37
Process
Next Sample In Index

Fig 9

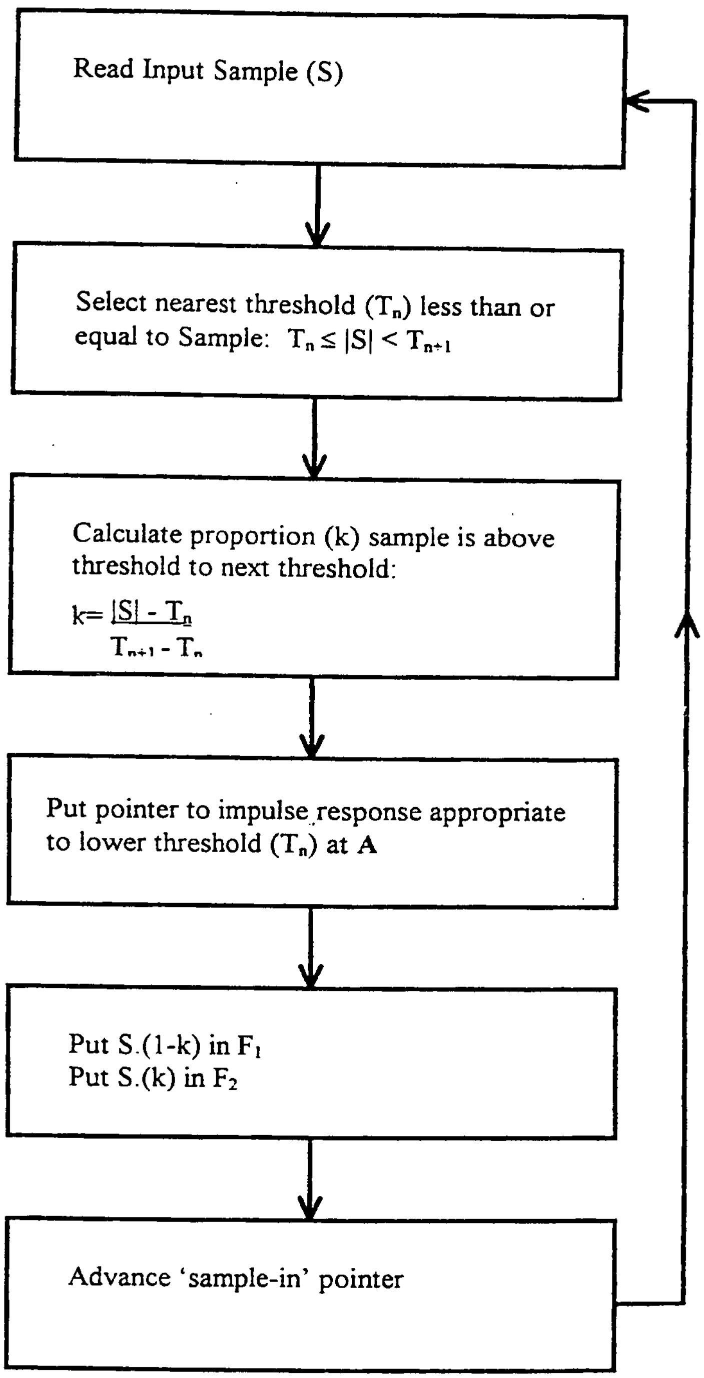

Read Input Sample (S)
Select nearest threshold ($T_n$) less than or equal to Sample: $T_n \leq |S| < T_{n+1}$
Calculate proportion (k) sample is above threshold to next threshold:
$k = \frac{|S| - T_n}{T_{n+1} - T_n}$
Put pointer to impulse response appropriate to lower threshold ($T_n$) at **A**
Put S.(1-k) in $F_1$
Put S.(k) in $F_2$
Advance 'sample-in' pointer

**Fig 10**

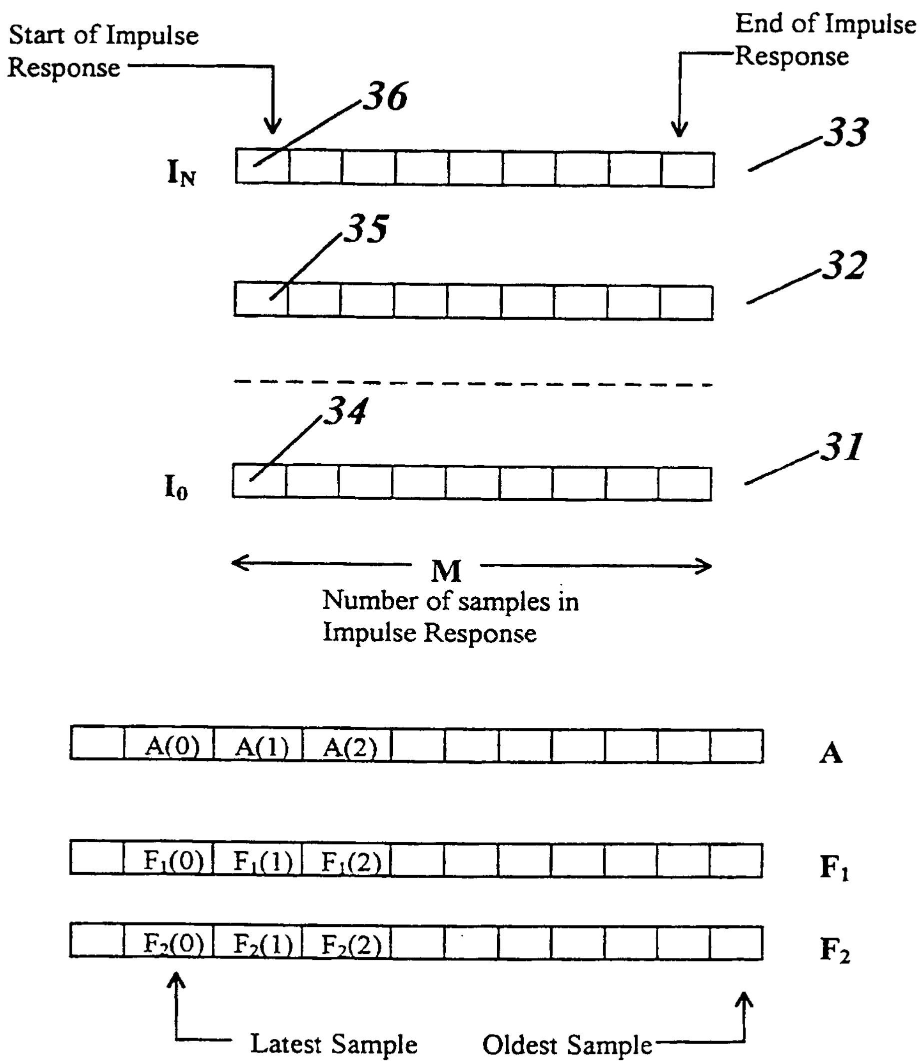

Start of Impulse Response
End of Impulse Response
36
33
$I_N$
35
32
34
31
$I_0$
M
Number of samples in Impulse Response
A(0) A(1) A(2)
A
$F_1(0)$ $F_1(1)$ $F_1(2)$
$F_1$
$F_2(0)$ $F_2(1)$ $F_2(2)$
$F_2$
Latest Sample
Oldest Sample

Fig 11

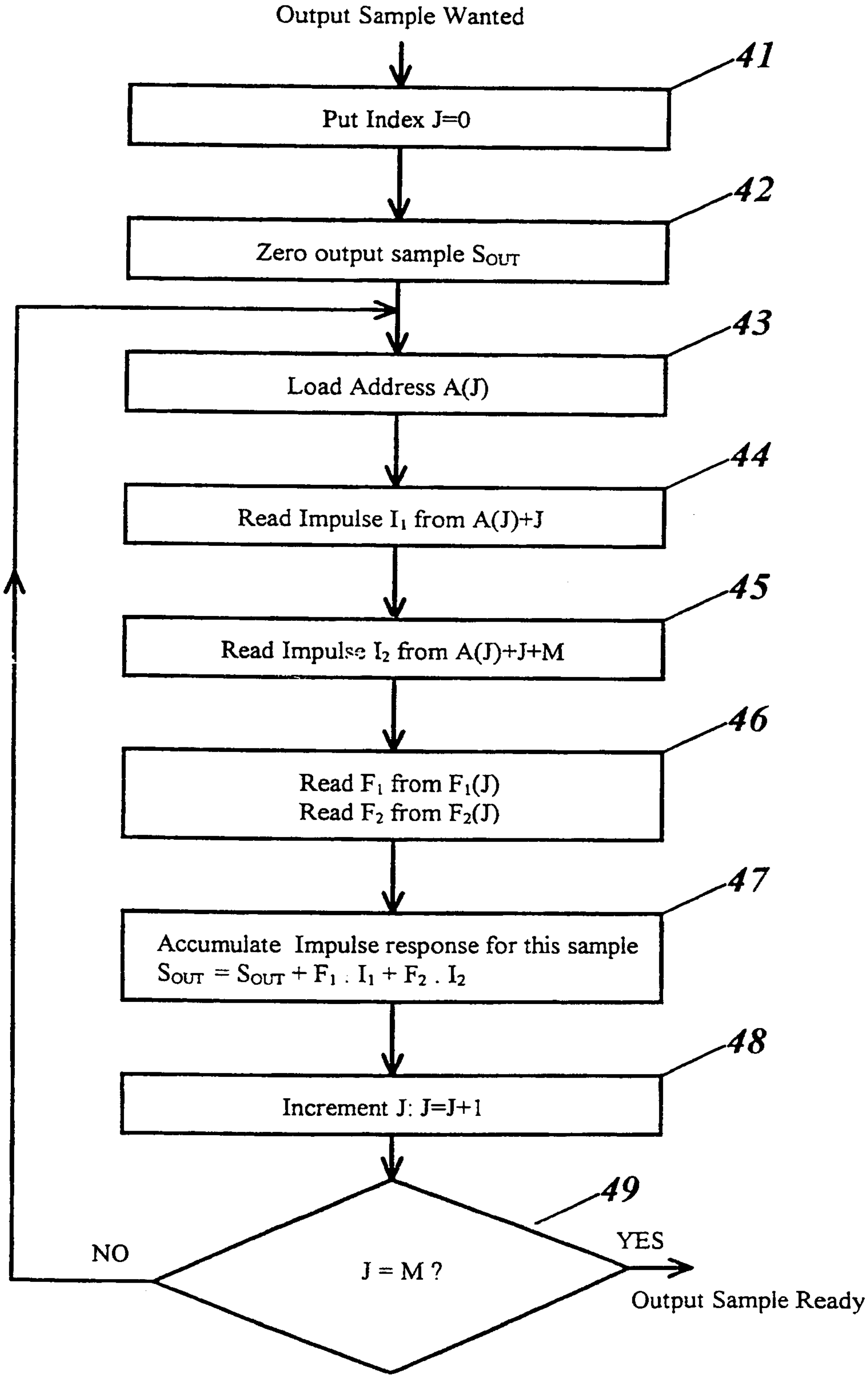
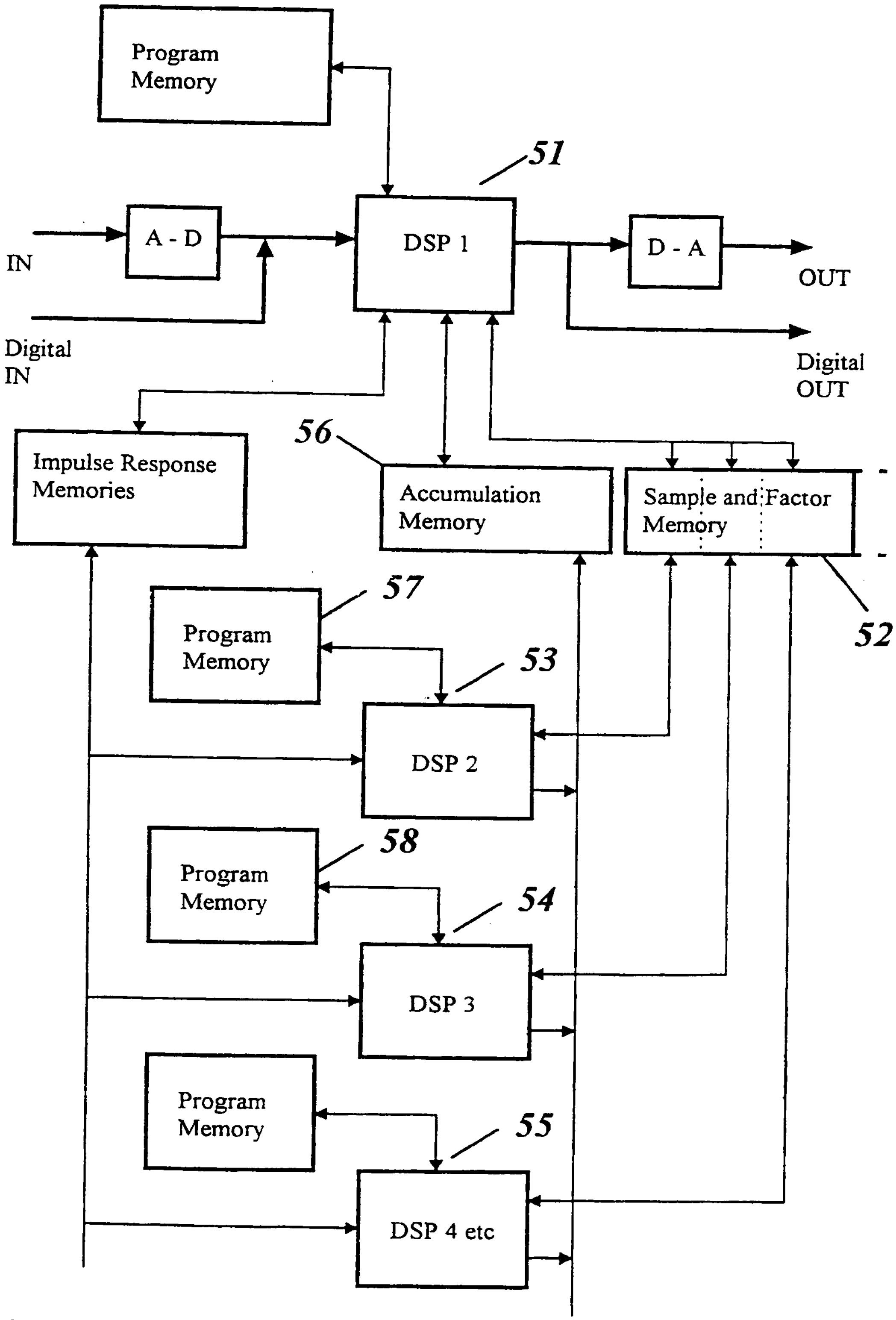

Output Sample Wanted
41
Put Index J=0
42
Zero output sample $S_{OUT}$
43
Load Address A(J)
44
Read Impulse $I_1$ from A(J)+J
45
Read Impulse $I_2$ from A(J)+J+M
46
Read $F_1$ from $F_1(J)$
Read $F_2$ from $F_2(J)$
47
Accumulate Impulse response for this sample
$S_{OUT} = S_{OUT} + F_1 . I_1 + F_2 . I_2$
48
Increment J: J=J+1
49
NO
J = M ?
YES
Output Sample Ready

Fig 12



Program Memory
51
IN
A - D
DSP 1
D - A
OUT
Digital IN
Digital OUT
56
Impulse Response Memories
Accumulation Memory
Sample and Factor Memory
52
57
Program Memory
53
DSP 2
58
Program Memory
54
DSP 3
Program Memory
55
DSP 4 etc


Fig 13

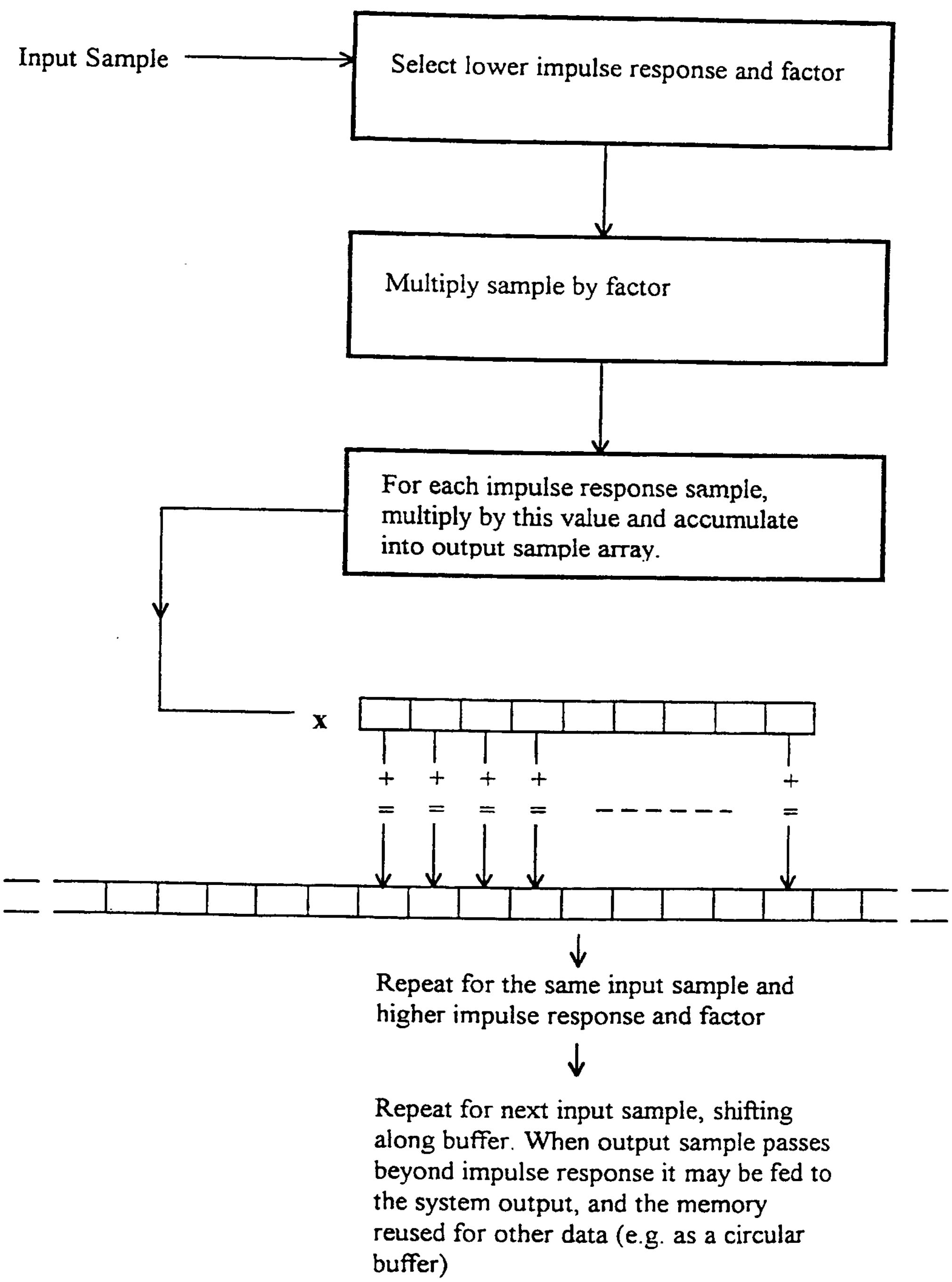

Input Sample
Select lower impulse response and factor
Multiply sample by factor
For each impulse response sample, multiply by this value and accumulate into output sample array.
x
Repeat for the same input sample and higher impulse response and factor
Repeat for next input sample, shifting along buffer. When output sample passes beyond impulse response it may be fed to the system output, and the memory reused for other data (e.g. as a circular buffer)

Fig 14

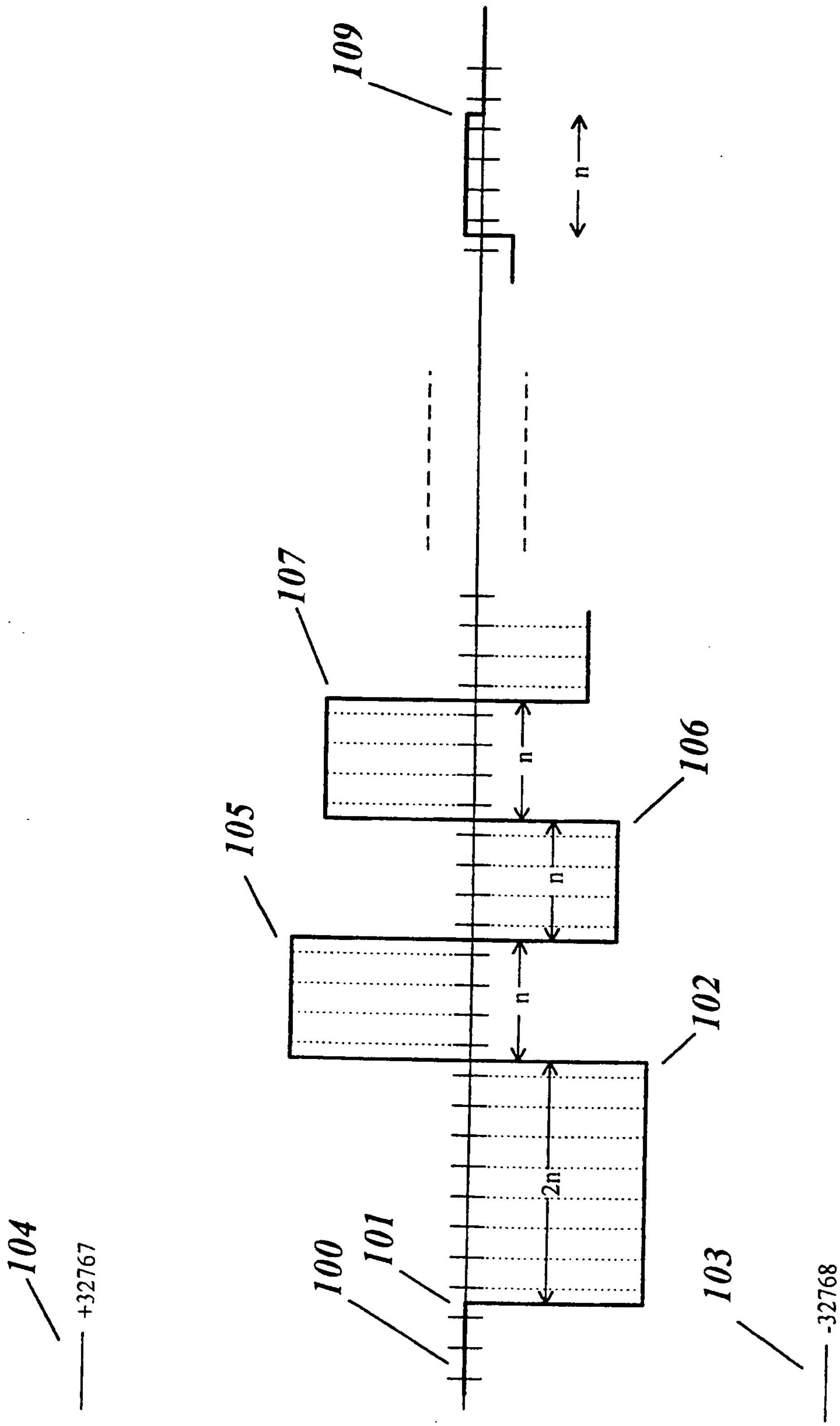



Fig 15

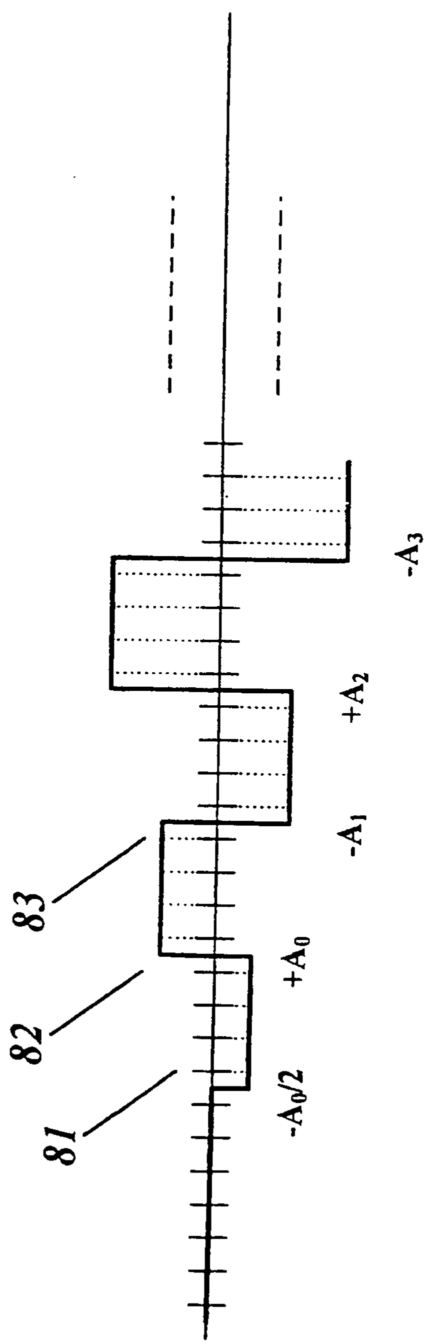



Fig 16

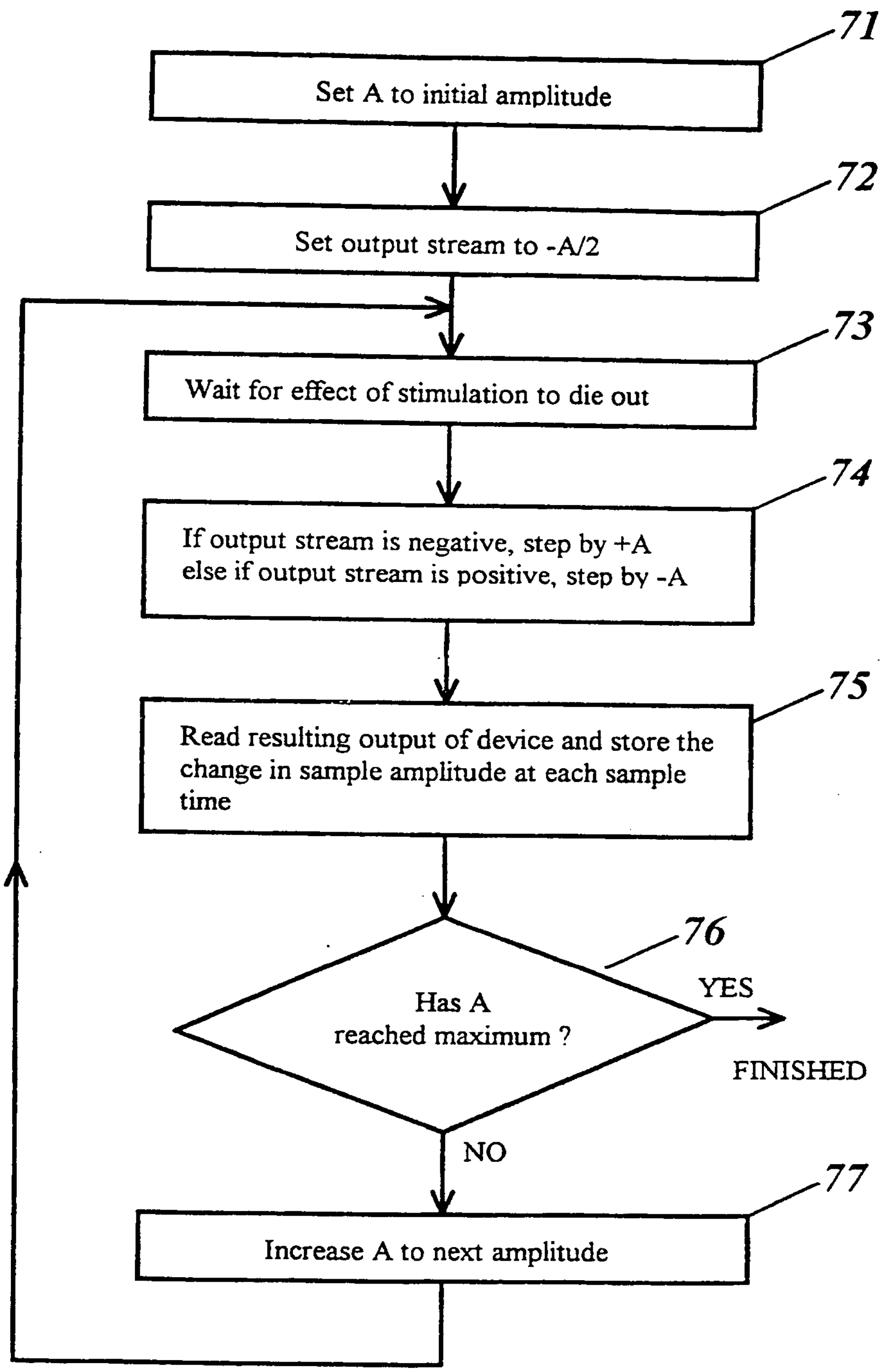

71
Set A to initial amplitude
72
Set output stream to -A/2
73
Wait for effect of stimulation to die out
74
If output stream is negative, step by +A
else if output stream is positive, step by -A
75
Read resulting output of device and store the change in sample amplitude at each sample time
76
Has A reached maximum ?
YES
FINISHED
NO
77
Increase A to next amplitude

Fig 17

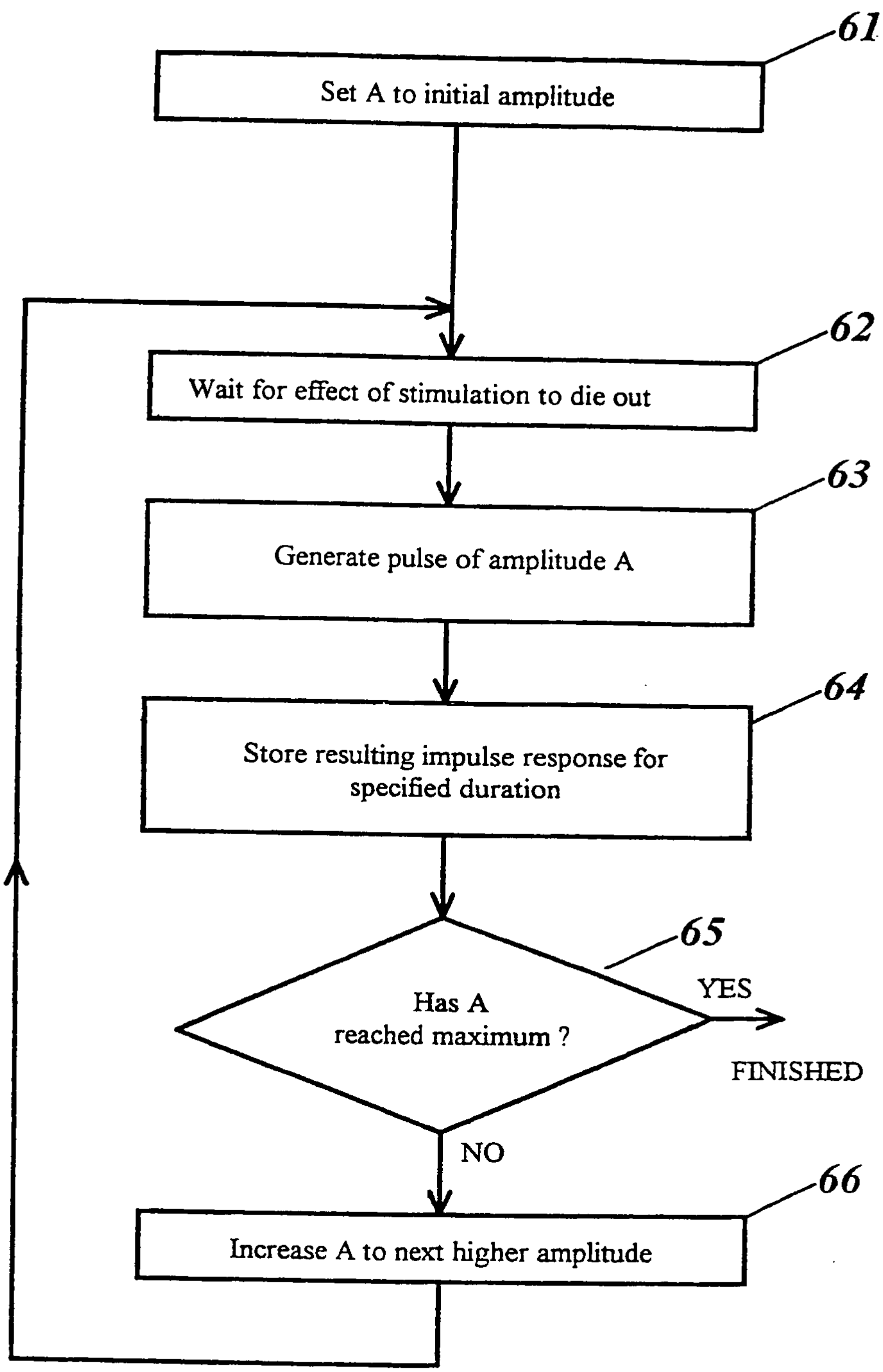

61
Set A to initial amplitude
62
Wait for effect of stimulation to die out
63
Generate pulse of amplitude A
64
Store resulting impulse response for specified duration
65
Has A reached maximum ?
YES
FINISHED
NO
66
Increase A to next higher amplitude

Fig 18

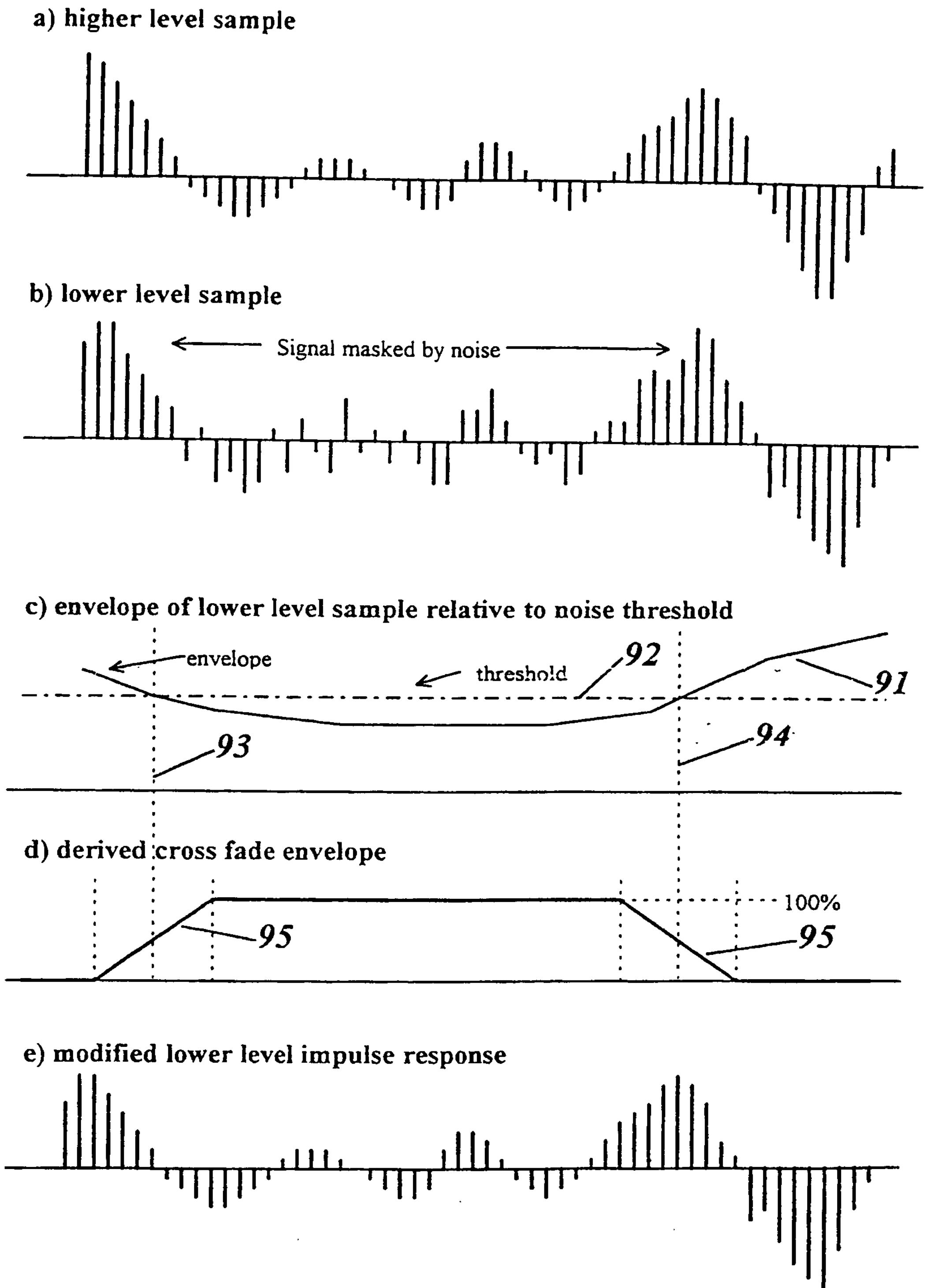

a) higher level sample
b) lower level sample
Signal masked by noise
c) envelope of lower level sample relative to noise threshold
envelope
threshold
92
91
93
94
d) derived cross fade envelope
100%
95
95
e) modified lower level impulse response

Fig 19

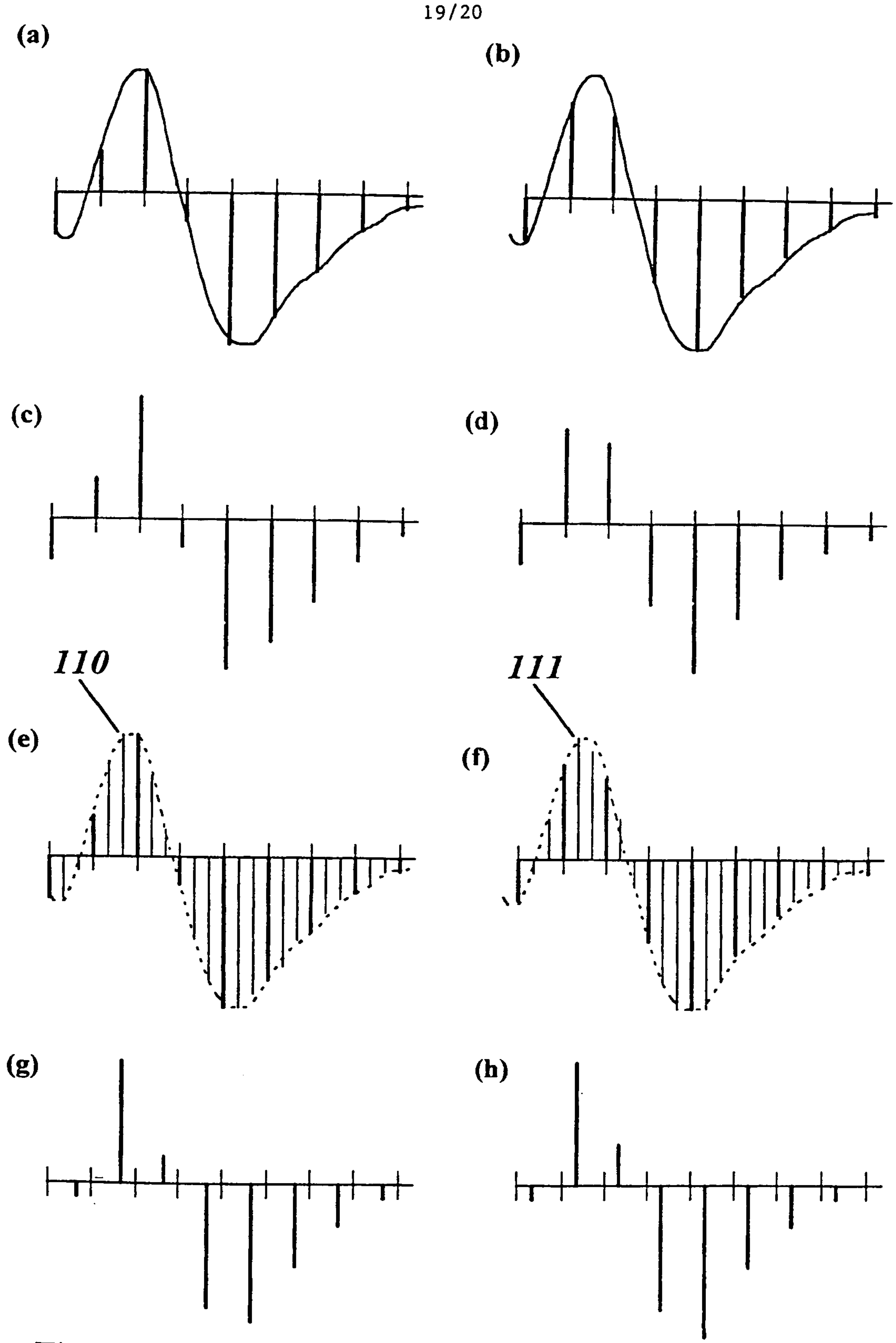



Fig 20

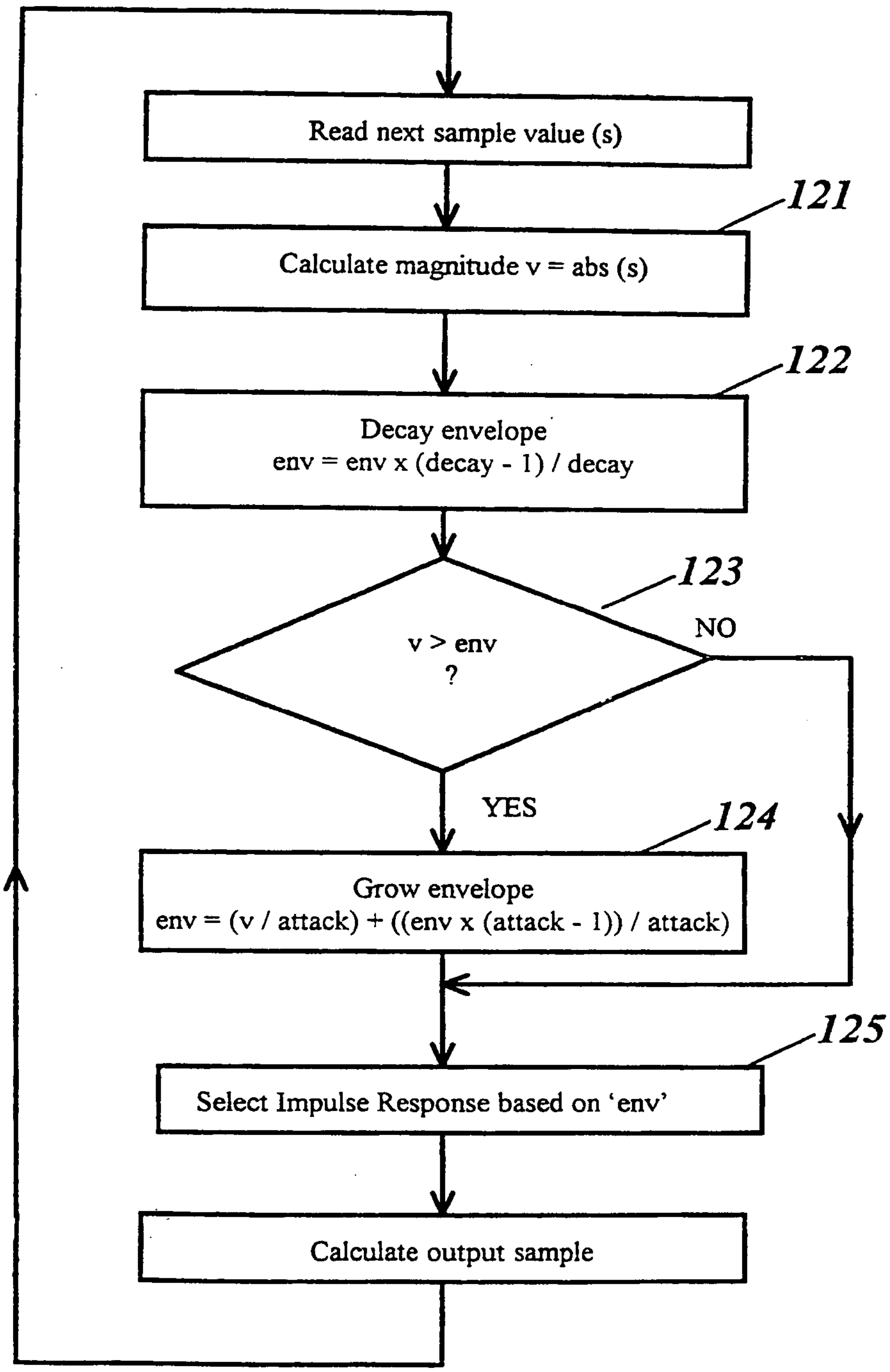

Read next sample value (s)
121
Calculate magnitude v = abs (s)
122
Decay envelope
env = env x (decay - 1) / decay
123
v > env
?
NO
YES
124
Grow envelope
env = (v / attack) + ((env x (attack - 1)) / attack)
125
Select Impulse Response based on 'env'
Calculate output sample

Fig 21

# INTERNATIONAL SEARCH REPORT

International Application No
PCT/GB 97/02159

**A. CLASSIFICATION OF SUBJECT MATTER**
IPC 6 G10K15/02

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 G10K

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

| Category ° | Citation of document, with indication, where appropriate, of the relevant passages | Relevant to claim No. |
|---|---|---|
| A | US 5 544 249 A (OPITZ MARTIN) 6 August 1996<br>see claim 1<br>----- | 1,3,4,<br>10,12,13 |

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

1

| Date of the actual completion of the international search | Date of mailing of the international search report |
|---|---|
| 15 December 1997 | 23/12/1997 |

| Name and mailing address of the ISA | Authorized officer |
|---|---|
| European Patent Office, P.B. 5818 Patentlaan 2<br>NL - 2280 HV Rijswijk<br>Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,<br>Fax: (+31-70) 340-3016 | Anderson, A |

# INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/GB 97/02159

| Patent document cited in search report | Publication date | Patent family member(s) | Publication date |
|---|---|---|---|
| US 5544249 A | 06-08-96 | DE 4328620 C | 19-01-95 |
| | | EP 0641143 A | 01-03-95 |
| | | JP 7087589 A | 31-03-95 |

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# PATENT COOPERATION TREATY

From the
INTERNATIONAL PRELIMINARY EXAMINING AUTHORITY

11 MAY 1998

To:

ROBSON, Aidan J.
REDDIE & GROSE
16, Theobalds Road
London WC1X 8PL
GRANDE BRETAGNE

3 mths
7.8.98
CHECKED
A.R.G.

# PCT

## WRITTEN OPINION

(PCT Rule 66)

| | | |
|---|---|---|
| | Date of mailing (day/month/year) | -7. 05. 98 |
| Applicant's or agent's file reference AJR/37836 | REPLY DUE | within 3 month(s) from the above date of mailing |

| International application no. | International filing date (day/month/year) | Priority date (day/month/year) |
|---|---|---|
| PCT/GB97/02159 | 08/08/1997 | 09/08/1996 |

International Patent Classification (IPC) or both national classification and IPC
G10K15/02

Applicant
KEMP, Michael Joseph

1. This written opinion is the **first** drawn up by this International Preliminary Examining Authority.

2. This report contains indications relating to the following items:
   - I ☒ Basis of the opinion
   - II ☐ Priority
   - III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   - IV ☐ Lack of unity of invention
   - V ☒ Reasoned statement under Rule 66.2(a)(ii) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement
   - VI ☐ Certain documents cited
   - VII ☒ Certain defects in the international application
   - VIII ☒ Certain observations on the international application

3. The applicant is hereby **invited to reply** to this opinion.

   **When?** See the time limit indicated above. The applicant may, before the expiration of that time limit, request this Authority to grant an extension, see Rule 66.2(d).

   **How?** By submitting a written reply, accompanied, where appropriate, by amendments, according to Rule 66.3. For the form and the language of the amendments, see Rules 66.8 and 66.9.

   **Also:** For an additional opportunity to submit amendments, see Rule 66.4.
   For the examiner's obligation to consider amendments and / or arguments, see Rule 66.4bis.
   For an informal communication with the examiner, see Rule 66.6.

   **If no reply is filed,** the international preliminary examination report will be established on the basis of this opinion.

4. The final date by which the international preliminary examination report must be established according to Rule 69.2 is: 09/12/1998

| Name and mailing address of the international preliminary examining authority | Authorized officer / Examiner |
|---|---|
| European Patent Office<br>D-80298 Munich<br>Tel. (+49-89) 2399-0, Tx: 523656 epmu d<br>Fax: (+49-89) 2399-4465 | Zwicker, T<br>Formalities officer (incl. extension of time limits)<br>Bapisch, A<br>Telephone No. (+49-89) 2399-2262 |

**I. Basis of the opinion**

1. This opinion has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this opinion as "originally filed".)*:

**Description, pages:**

| | |
|---|---|
| 1-32 | as originally filed |

**Claims, No.:**

| | |
|---|---|
| 1-22 | as originally filed |

**Drawings, sheets:**

| | |
|---|---|
| 1/20-20/20 | as originally filed |

2. The amendments have resulted in the cancellation of:

- ☐ the description, pages:
- ☐ the claims, Nos.:
- ☐ the drawings, sheets:

3. This opinion has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

**V. Reasoned statement under Rule 66.2(a)(ii) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement**

1. Statement

| | | |
|---|---|---|
| Novelty (N) | Claims | 1-20 (yes) |
| Inventive step (IS) | Claims | 1-20 (yes) |
| Industrial applicability (IA) | Claims | 1-20 (yes) |

2. Citations and explanations

**see separate sheet**

### VII. Certain defects in the international application

The following defects in the form or contents of the international application have been noted:

**see separate sheet**

### VIII. Certain observations on the international application

The following observations on the clarity of the claims, description, and drawings or on the question whether the claims are fully supported by the description, are made:

**see separate sheet**

**V. With respect to section V of this opinion**

1. It would appear that the presently available state of the art does not disclose or render obvious the subject matter of present independent claims 1 (methos) and 10 (corresponding apparatus). Therefore these claims appear to meet the requirements of Art. 33 PCT.

2. Present dependent claims 2 - 9 and 11 - 20 represent advantageous embodiments of the basic principles of claims 1 and 10, and would therefore also appear to meet the requirements of Art. 33 PCT.

**VII. With respect to section VII of this opinion**

1. Although it is acknowledged that document D1 (US-A-5 544 249) is, due to the lack of the feature of choosing an impulse response (as found in present claims 1 and 10), not particularly relevant to the present application, it should, in view of Rule 5.1(a)(ii) PCT, still be mentioned and identified as background art in the description.

2. The features of the claims are not provided with reference signs placed in parentheses (Rule 6.2(b) PCT).

**VIII. With respect to section VIII of this opinion**

Present claims 21 and 22 do not met the requirements of Art. 6 PCT because they are not clear, because the wording of these claims requires them to interpreted as:
"A method (apparatus) **suitable** for storing the impulse response of an audio processor for use in the method (apparatus) of any of claims 1-9 (10-18)."

However, this means that any apparatus (method) suitable for this task would be claimed, with no specific features at all defined for this method (apparatus). Since claims 21 and 22 consequently do not hold any features which would be

appropriate for defining an invention, these claims are not clear.

At present it appears that claims 21 and 22 should be deleted.

EPA/EPO/OEB
D-80298 München
(089) 2399-0
TX 523 656 epmu d
FAX (089) 2399-4465



## Correspondence with the EPO on PCT Chapter II demands

In order to ensure that your PCT Chapter II demand is dealt with as promptly as possible you are requested to use the enclosed self-adhesive labels with any correspondence relating to the demand sent to the Munich Office.

One of these labels should be affixed to a prominent place in the upper part of the letter or form etc. which you are filing.

19

REC'L 09 SEP 1998
WIPO PCT

# PATENT COOPERATION TREATY

# PCT

## INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

| | | |
|---|---|---|
| Applicant's or agent's file reference<br>AJR/37836 | **FOR FURTHER ACTION** | See Notification of Transmittal of International Preliminary Examination Report (PCT/IPEA/416) |
| International application No.<br>PCT/GB97/02159 | International filing date *(day/month/year)*<br>08/08/1997 | Priority date *(day/month/year)*<br>09/08/1996 |
| International Patent Classification (IPC) or national classification and IPC<br>G10K15/02 | | |
| Applicant<br>KEMP, Michael Joseph | | |

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 5 sheets, including this cover sheet.

   ☐ This report is also accompanied by ANNEXES, i.e., sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

   These annexes consist of a total of sheets.

3. This report contains indications relating to the following items:

   I ☒ Basis of the report
   II ☐ Priority
   III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   IV ☐ Lack of unity of invention
   V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement
   VI ☐ Certain documents cited
   VII ☒ Certain defects in the international application
   VIII ☒ Certain observations on the international application

| | |
|---|---|
| Date of submission of the demand<br>27/02/1998 | Date of completion of this report<br>07.09.98 |
| Name and mailing address of the IPEA/<br>European Patent Office<br>D-80298 Munich<br>Tel. (+49-89) 2399-0, Tx: 523656 epmu d<br>Fax: (+49-89) 2399-4465 | Authorized officer<br>Zwicker, T<br>Telephone No. (+49-89) 2399-2841 |

Form PCT/IPEA/409 (Cover sheet) (January 1994)

**I. Basis of the report**

1. This report has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to the report since they do not contain amendments.*):

**Description, pages:**

1-32 as originally filed

**Claims, No.:**

1-22 as originally filed

**Drawings, sheets:**

1/20-20/20 as originally filed

2. The amendments have resulted in the cancellation of:

☐ the description, pages:
☐ the claims, Nos.:
☐ the drawings, sheets:

3. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

**V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement**

1. Statement

| | | | |
|---|---|---|---|
| Novelty (N) | Yes: | Claims | 1-20 |
| | No: | Claims | 21,22 |
| Inventive step (IS) | Yes: | Claims | 1-20 |
| | No: | Claims | 21,22 |
| Industrial applicability (IA) | Yes: | Claims | 1-22 |
| | No: | Claims | |

2. Citations and explanations

**see separate sheet**

**VII. Certain defects in the international application**

The following defects in the form or contents of the international application have been noted:

**see separate sheet**

**VIII. Certain observations on the international application**

The following observations on the clarity of the claims, description, and drawings or on the question whether the claims are fully supported by the description, are made:

**see separate sheet**

## V. With respect to section V of this report

V.1 The present application relates to a device for audio effects.

Document D1 (US-A-5 544 249) cited in the search report and considered to be the closest state of the art discloses an effects processor for simulating reverb and room acoustics in listening environments. The impulse response for a specific listening environment is determined, and those input signals to the processor which have at least a specific predetermined amplitude subjected to this impulse response. With this approach, less computing power is necessary, and often a cleaner sound achieved.

The present application seeks to provide a very variable simulation of audio effects. To this end, claims 1 (method) and 10 (corresponding apparatus) propose to store the impulse response of an audio processor for at least two impulse responses, and subject an input signal on a selected one of these impulse responses. The selection is made dependent on a characteristic (e.g. signal level) of the input signal.

Using this approach, specific properties (e.g. non-linearities) of a signal processor can effectively be simulated.

The presently available state of the art does not point in any way to the features of assessing more than one impulse response and selecting an impulse response dependent on an input signal characteristic.

Consequently, the subject matter of present independent claims 1 and 10 is not obvious, and the claims would appear to meet the requirements of Art. 33 PCT.

Dependent claims 2 - 9 and 11 - 20 represent advantageous embodiments of the basic concept of claims 1 and 2, and would therefore also appear to meet the requirements of Art. 33 PCT.

V.2 Claims 21 and 22, as far as they are understood in view of the below section VIII do not meet the requirements of Art. 33 PCT because they are not novel.

This is because the wording of the claim 21 (22 requires it to interpreted as: "A method (apparatus) **suitable** for storing the impulse response of an audio processor for use in the method (apparatus) of any of claims 1-9 (10-18)."

However, this means that any apparatus (method) suitable for this task (e.g. any computer with AD- and DA-conversion) would fit the wording of claim 21 (22). Claims 21 and 22 can consequently not be considered novel.

## VII. With respect to section VII of this opinion

1. Although it is acknowledged that document D1 (US-A-5 544 249) is, due to the lack of the feature of choosing an impulse response (as found in present claims 1 and 10), not particularly relevant to the present application, it should, in view of Rule 5.1(a)(ii) PCT, still have been mentioned and identified as background art in the description.

2. The features of the claims are not provided with reference signs placed in parentheses (Rule 6.2(b) PCT).

## VIII. With respect to section VIII of this opinion

Present claims 21 and 22 do not met the requirements of Art. 6 PCT because they are not clear, since the wording of these claims requires them to interpreted as: "A method (apparatus) **suitable** for storing the impulse response of an audio processor for use in the method (apparatus) of any of claims 1-9 (10-18)."

However, this means that any apparatus (method) suitable for this task would be claimed, with no specific features at all defined for this method (apparatus). Since claims 21 and 22 consequently do not hold any features which would be appropriate for defining an invention, these claims are not clear.

It appears, in particular in view of the above section V.2, that claims 21 and 22 should have been deleted.

**PATENT COOPERATION TREATY**

09 SEP 1998

From the
INTERNATIONAL PRELIMINARY EXAMINING AUTHORITY

To:

ROBSON, Aidan J.
REDDIE & GROSE
16, Theobalds Road
London WC1X 8PL
GRANDE BRETAGNE

# PCT

## NOTIFICATION OF TRANSMITTAL OF THE INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Rule 71.1)

9.2.99

| | |
|---|---|
| Date of mailing (day/month/year) | 07.09.98 |
| Applicant's or agent's file reference AJR/37836 | **IMPORTANT NOTIFICATION** |

| International application No. | International filing date (day/month/year) | Priority date (day/month/year) |
|---|---|---|
| PCT/GB97/02159 | 08/08/1997 | 09/08/1996 |

Applicant
KEMP, Michael Joseph

1. The applicant is hereby notified that this International Preliminary Examining Authority transmits herewith the international preliminary examination report and its annexes, if any, established on the international application.

2. A copy of the report and its annexes, if any, is being transmitted to the International Bureau for communication to all the elected Offices.

3. Where required by any of the elected Offices, the International Bureau will prepare an English translation of the report (but not of any annexes) and will transmit such translation to those Offices.

4. **REMINDER**

The applicant must enter the national phase before each elected Office by performing certain acts (filing translations and paying national fees) within 30 months from the priority date (or later in some Offices) (Article 39(1)) (see also the reminder sent by the International Bureau with Form PCT/IB/301).

Where a translation of the international application must be furnished to an elected Office, that translation must contain a translation of any annexes to the international preliminary examination report. It is the applicant's responsibility to prepare and furnish such translation directly to each elected Office concerned.

For further details on the applicable time limits and requirements of the elected Offices, see Volume II of the PCT Applicant's Guide.

| Name and mailing address of the IPEA/ | Authorized officer |
|---|---|
| European Patent Office<br>D-80298 Munich<br>Tel. (+49-89) 2399-0, Tx: 523656 epmu d<br>Fax: (+49-89) 2399-4465 | Swartebroeckx, J-J<br>Tel. (+49-89) 2399-2692 |

# PATENT COOPERATION TREATY

# PCT

## INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

| Applicant's or agent's file reference<br>AJR/37836 | **FOR FURTHER ACTION** | See Notification of Transmittal of International Preliminary Examination Report (PCT/IPEA/416) |
|---|---|---|
| International application No.<br>PCT/GB97/02159 | International filing date *(day/month/year)*<br>08/08/1997 | Priority date *(day/month/year)*<br>09/08/1996 |
| International Patent Classification (IPC) or national classification and IPC<br>G10K15/02 | | |
| Applicant<br>KEMP, Michael Joseph | | |

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 5 sheets, including this cover sheet.

   ☐ This report is also accompanied by ANNEXES, i.e., sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

   These annexes consist of a total of sheets.

3. This report contains indications relating to the following items:

   I ☒ Basis of the report
   II ☐ Priority
   III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   IV ☐ Lack of unity of invention
   V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement
   VI ☐ Certain documents cited
   VII ☒ Certain defects in the international application
   VIII ☒ Certain observations on the international application

| Date of submission of the demand | Date of completion of this report |
|---|---|
| 27/02/1998 | 07.09.98 |
| Name and mailing address of the IPEA/<br>European Patent Office<br>D-80298 Munich<br>Tel. (+49-89) 2399-0, Tx: 523656 epmu d<br>Fax: (+49-89) 2399-4465 | Authorized officer<br>Zwicker, T<br>Telephone No. (+49-89) 2399-2841 |

**INTERNATIONAL PRELIMINARY EXAMINATION REPORT**

International application No. PCT/GB97/02159

## I. Basis of the report

1. This report has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to the report since they do not contain amendments.*):

**Description, pages:**

1-32 as originally filed

**Claims, No.:**

1-22 as originally filed

**Drawings, sheets:**

1/20-20/20 as originally filed

2. The amendments have resulted in the cancellation of:

- ☐ the description, pages:
- ☐ the claims, Nos.:
- ☐ the drawings, sheets:

3. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

**V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement**

1. Statement

| | | | |
|---|---|---|---|
| Novelty (N) | Yes: | Claims | 1-20 |
| | No: | Claims | 21,22 |
| Inventive step (IS) | Yes: | Claims | 1-20 |
| | No: | Claims | 21,22 |
| Industrial applicability (IA) | Yes: | Claims | 1-22 |
| | No: | Claims | |

2. Citations and explanations

**see separate sheet**

**VII. Certain defects in the international application**

The following defects in the form or contents of the international application have been noted:

**see separate sheet**

**VIII. Certain observations on the international application**

The following observations on the clarity of the claims, description, and drawings or on the question whether the claims are fully supported by the description, are made:

**see separate sheet**

## V. With respect to section V of this report

V.1 The present application relates to a device for audio effects.

Document D1 (US-A-5 544 249) cited in the search report and considered to be the closest state of the art discloses an effects processor for simulating reverb and room acoustics in listening environments. The impulse response for a specific listening environment is determined, and those input signals to the processor which have at least a specific predetermined amplitude subjected to this impulse response. With this approach, less computing power is necessary, and often a cleaner sound achieved.

The present application seeks to provide a very variable simulation of audio effects. To this end, claims 1 (method) and 10 (corresponding apparatus) propose to store the impulse response of an audio processor for at least two impulse responses, and subject an input signal on a selected one of these impulse responses. The selection is made dependent on a characteristic (e.g. signal level) of the input signal.

Using this approach, specific properties (e.g. non-linearities) of a signal processor can effectively be simulated.

The presently available state of the art does not point in any way to the features of assessing more than one impulse response and selecting an impulse response dependent on an input signal characteristic.

Consequently, the subject matter of present independent claims 1 and 10 is not obvious, and the claims would appear to meet the requirements of Art. 33 PCT.

Dependent claims 2 - 9 and 11 - 20 represent advantageous embodiments of the basic concept of claims 1 and 2, and would therefore also appear to meet the requirements of Art. 33 PCT.

V.2 Claims 21 and 22, as far as they are understood in view of the below section VIII do not meet the requirements of Art. 33 PCT because they are not novel.

This is because the wording of the claim 21 (22 requires it to interpreted as: "A method (apparatus) **suitable** for storing the impulse response of an audio processor for use in the method (apparatus) of any of claims 1-9 (10-18)."

However, this means that any apparatus (method) suitable for this task (e.g. any computer with AD- and DA-conversion) would fit the wording of claim 21 (22). Claims 21 and 22 can consequently not be considered novel.

## VII. With respect to section VII of this opinion

1. Although it is acknowledged that document D1 (US-A-5 544 249) is, due to the lack of the feature of choosing an impulse response (as found in present claims 1 and 10), not particularly relevant to the present application, it should, in view of Rule 5.1(a)(ii) PCT, still have been mentioned and identified as background art in the description.

2. The features of the claims are not provided with reference signs placed in parentheses (Rule 6.2(b) PCT).

## VIII. With respect to section VIII of this opinion

Present claims 21 and 22 do not met the requirements of Art. 6 PCT because they are not clear, since the wording of these claims requires them to interpreted as: "A method (apparatus) **suitable** for storing the impulse response of an audio processor for use in the method (apparatus) of any of claims 1-9 (10-18)."

However, this means that any apparatus (method) suitable for this task would be claimed, with no specific features at all defined for this method (apparatus). Since claims 21 and 22 consequently do not hold any features which would be appropriate for defining an invention, these claims are not clear.

It appears, in particular in view of the above section V.2, that claims 21 and 22 should have been deleted.

**PATENT COOPERATION TREATY**

RECEIVED 29 DEC 1997

From the INTERNATIONAL SEARCHING AUTHORITY

**PCT**

| To: | VISTEM | TERM: | NOTIFICATION OF TRANSMITTAL OF THE INTERNATIONAL SEARCH REPORT OR THE DECLARATION |
|---|---|---|---|
| REDDIE & GROSE<br>Attn. ROBSON, Aidan J.<br>16, Theobalds Road<br>LONDON WC1X 8PL<br>UNITED KINGDOM | TECHNICAL AJR<br>EUROPEAN<br>FOREIGNS<br>REGISTERS<br>A.F.S. | 2 mths<br>DATE: 23/2/98<br>INITIALS / CHECKED: MP | (PCT Rule 44.1) |

| | |
|---|---|
| | Date of mailing (*day/month/year*) 23/12/1997 |
| Applicant's or agent's file reference AJR/37836 | **FOR FURTHER ACTION** See paragraphs 1 and 4 below |
| International application No. PCT/GB 97/ 02159 | International filing date (*day/month/year*) 08/08/1997 |
| Applicant KEMP, Michael Joseph | |

1. [X] The applicant is hereby notified that the International Search Report has been established and is transmitted herewith.

   **Filing of amendments and statement under Article 19**
   The applicant is entitled, if he so wishes, to amend the claims of the International Application (see Rule 46):

   **When?** The time limit for filing such amendments is normally 2 months from the date of transmittal of the International Search Report; however, for more details, see the notes on the accompanying sheet.

   **Where?** Directly to the International Bureau of WIPO
   34, chemin des Colombettes
   1211 Geneva 20, Switzerland
   Fascimile No.: (41-22) 740.14.35

   **For more detailed instructions,** see the notes on the accompanying sheet.

2. [ ] The applicant is hereby notified that no International Search Report will be established and that the declaration under Article 17(2)(a) to that effect is transmitted herewith.

3. [ ] **With regard to the protest** against payment of (an) additional fee(s) under Rule 40.2, the applicant is notified that:

   [ ] the protest together with the decision thereon has been transmitted to the International Bureau together with the applicants's request to forward the texts of both the protest and the decision thereon to the designated Offices.

   [ ] no decision has been made yet on the protest; the applicant will be notified as soon as a decision is made.

4. **Further action(s):** The applicant is reminded of the following:

   Shortly after **18 months** from the priority date, the international application will be published by the International Bureau. If the applicant wishes to avoid or postpone publication, a notice of withdrawal of the international application, or of the priority claim, must reach the International Bureau as provided in Rules 90*bis*.1 and 90*bis*.3, respectively, before the completion of the technical preparations for international publication.

   Within **19 months** from the priority date, a demand for international preliminary examination must be filed if the applicant wishes to postpone the entry into the national phase until 30 months from the priority date (in some Offices even later).

   Within **20 months** from the priority date, the applicant must perform the prescribed acts for entry into the national phase before all designated Offices which have not been elected in the demand or in a later election within 19 months from the priority date or could not be elected because they are not bound by Chapter II.

| Name and mailing address of the International Searching Authority | Authorized officer |
|---|---|
| European Patent Office, P.B. 5818 Patentlaan 2<br>NL-2280 HV Rijswijk<br>Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,<br>Fax: (+31-70) 340-3016 | Marie-Françoise Provot |

## NOTES TO FORM PCT/ISA/220

These Notes are intended to give the basic instructions concerning the filing of amendments under article 19. The Notes are based on the requirements of the Patent Cooperation Treaty, the Regulations and the Administrative Instructions under that Treaty. In case of discrepancy between these Notes and those requirements, the latter are applicable. For more detailed information, see also the PCT Applicant's Guide, a publication of WIPO.

In these Notes, "Article", "Rule", and "Section" refer to the provisions of the PCT, the PCT Regulations and the PCT Administrative Instructions respectively.

## INSTRUCTIONS CONCERNING AMENDMENTS UNDER ARTICLE 19

The applicant has, after having received the international search report, one opportunity to amend the claims of the international application. It should however be emphasized that, since all parts of the international application (claims, description and drawings) may be amended during the international preliminary examination procedure, there is usually no need to file amendments of the claims under Article 19 except where, e.g. the applicant wants the latter to be published for the purposes of provisional protection or has another reason for amending the claims before international pbulication. Furthermore, it should be emphasized that provisional protection is available in some States only.

**What parts of the international application may be amended?**

Under Article 19, only the claims may be amended.

During the international phase, the claims may also be amended (or further amended) under Article 34 before the International Preliminary Examining Authority. The description and drawings may only be amended under Article 34 before the International Examining Authority.

Upon entry into the national phase, all parts of the international application may be amended under Article 28 or, where applicable, Article 41.

**When?** Within 2 months from the date of transmittal of the international search report or 16 months from the priority date, whichever time limit expires later. It should be noted, however, that the amendments will be considered as having been received on time if they are received by the International Bureau after the expiration of the applicable time limit but before the completion of the technical preparations for international publication (Rule 46.1).

**Where not to file the amendments?**

The amendments may only be filed with the International Bureau and not with the receiving Office or the International Searching Authority (Rule 46.2).

Where a demand for international preliminary examination has been/is filed, see below.

**How?** Either by cancelling one or more entire claims, by adding one or more new claims or by amending the text of one or more of the claims as filed.

A replacement sheet must be submitted for each sheet of the claims which, on account of an amendment or amendments, differs from the sheet originally filed.

All the claims appearing on a replacement sheet must be numbered in Arabic numerals. Where a claim is cancelled, no renumbering of the other claims is required. In all cases where claims are renumbered, they must be renumbered consecutively (Administrative Instructions, Section 205(b)).

**The amendments must be made in the language in which the international application is to be published.**

**What documents must/may accompany the amendments?**

**Letter (Section 205(b)):**

The amendments must be submitted with a letter.

The letter will not be published with the international application and the amended claims. It should not be confused with the "Statement under Article 19(1)" (see below, under "Statement under Article 19(1)").

**The letter must be in English or French, at the choice of the applicant. However, if the language of the international application is English, the letter must be in English; if the language of the international application is French, the letter must be in French.**

## NOTES TO FORM PCT/ISA/220 (continued)

The letter must indicate the differences between the claims as filed and the claims as amended. It must, in particular, indicate, in connection with each claim appearing in the international application (it being understood that identical indications concerning several claims may be grouped), whether

(i) the claim is unchanged;

(ii) the claim is cancelled;

(iii) the claim is new;

(iv) the claim replaces one or more claims as filed;

(v) the claim is the result of the division of a claim as filed.

**The following examples illustrate the manner in which amendments must be explained in the accompanying letter:**

1. [Where originally there were 48 claims and after amendment of some claims there are 51]: "Claims 1 to 29, 31, 32, 34, 35, 37 to 48 replaced by amended claims bearing the same numbers; claims 30, 33 and 36 unchanged; new claims 49 to 51 added."
2. [Where originally there were 15 claims and after amendment of all claims there are 11]: "Claims 1 to 15 replaced by amended claims 1 to 11."
3. [Where originally there were 14 claims and the amendments consist in cancelling some claims and in adding new claims]: "Claims 1 to 6 and 14 unchanged; claims 7 to 13 cancelled; new claims 15, 16 and 17 added." or "Claims 7 to 13 cancelled; new claims 15, 16 and 17 added; all other claims unchanged."
4. [Where various kinds of amendments are made]: "Claims 1-10 unchanged; claims 11 to 13, 18 and 19 cancelled; claims 14, 15 and 16 replaced by amended claim 14; claim 17 subdivided into amended claims 15, 16 and 17; new claims 20 and 21 added."

**"Statement under article 19(1)" (Rule 46.4)**

The amendments may be accompanied by a statement explaining the amendments and indicating any impact that such amendments might have on the description and the drawings (which cannot be amended under Article 19(1)).

The statement will be published with the international application and the amended claims.

**It must be in the language in which the international appplication is to be published.**

It must be brief, not exceeding 500 words if in English or if translated into English.

It should not be confused with and does not replace the letter indicating the differences between the claims as filed and as amended. It must be filed on a separate sheet and must be identified as such by a heading, preferably by using the words "Statement under Article 19(1)."

It may not contain any disparaging comments on the international search report or the relevance of citations contained in that report. Reference to citations, relevant to a given claim, contained in the international search report may be made only in connection with an amendment of that claim.

**Consequence if a demand for international preliminary examination has already been filed**

If, at the time of filing any amendments under Article 19, a demand for international preliminary examination has already been submitted, the applicant must preferably, at the same time of filing the amendments with the International Bureau, also file a copy of such amendments with the International Preliminary Examining Authority (see Rule 62.2(a), first sentence).

**Consequence with regard to translation of the international application for entry into the national phase**

The applicant's attention is drawn to the fact that, where upon entry into the national phase, a translation of the claims as amended under Article 19 may have to be furnished to the designated/elected Offices, instead of, or in addition to, the translation of the claims as filed.

For further details on the requirements of each designated/elected Office, see Volume II of the PCT Applicant's Guide.

# PATENT COOPERATION TREATY

# PCT

## INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

| Applicant's or agent's file reference AJR/37836 | **FOR FURTHER ACTION** see Notification of Transmittal of International Search Report (Form PCT/ISA/220) as well as, where applicable, item 5 below. | |
|---|---|---|
| International application No. PCT/GB 97/02159 | International filing date *(day/month/year)* 08/08/1997 | (Earliest) Priority Date *(day/month/year)* 09/08/1996 |
| Applicant KEMP, Michael Joseph | | |

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 2 sheets.

[X] It is also accompanied by a copy of each prior art document cited in this report.

1. [ ] **Certain claims were found unsearchable** (see Box I).

2. [ ] **Unity of invention is lacking** (see Box II).

3. [ ] The international application contains disclosure of a **nucleotide and/or amino acid sequence listing** and the international search was carried out on the basis of the sequence listing
   - [ ] filed with the international application.
   - [ ] furnished by the applicant separately from the international application,
     - [ ] but not accompanied by a statement to the effect that it did not include matter going beyond the disclosure in the international application as filed.
   - [ ] Transcribed by this Authority

4. With regard to the **title**,
   - [X] the text is approved as submitted by the applicant
   - [ ] the text has been established by this Authority to read as follows:

5. With regard to the **abstract**,
   - [X] the text is approved as submitted by the applicant
   - [ ] the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this International Search Report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is:

   Figure No. 8
   - [ ] as suggested by the applicant.
   - [X] because the applicant failed to suggest a figure.
   - [ ] because this figure better characterizes the invention.
   - [ ] None of the figures.

# INTERNATIONAL SEARCH REPORT

International Application No
PCT/GB 97/02159

**A. CLASSIFICATION OF SUBJECT MATTER**
IPC 6 G10K15/02

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 G10K

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

| Category ° | Citation of document, with indication, where appropriate, of the relevant passages | Relevant to claim No. |
|---|---|---|
| A | US 5 544 249 A (OPITZ MARTIN) 6 August 1996<br>see claim 1<br>----- | 1,3,4,<br>10,12,13 |

[ ] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

| Date of the actual completion of the international search | Date of mailing of the international search report |
|---|---|
| 15 December 1997 | 23/12/1997 |

| Name and mailing address of the ISA | Authorized officer |
|---|---|
| European Patent Office, P.B. 5818 Patentlaan 2<br>NL – 2280 HV Rijswijk<br>Tel. (+31–70) 340–2040, Tx. 31 651 epo nl,<br>Fax: (+31–70) 340–3016 | Anderson, A |

1

# INTERNATIONAL SEARCH REPORT

**Information on patent family members**

International Application No
PCT/GB 97/02159

| Patent document cited in search report | Publication date | Patent family member(s) | Publication date |
|---|---|---|---|
| US 5544249 A | 06-08-96 | DE 4328620 C | 19-01-95 |
| | | EP 0641143 A | 01-03-95 |
| | | JP 7087589 A | 31-03-95 |

US005544249A

# United States Patent [19]

**Opitz**

[11] **Patent Number: 5,544,249**

[45] **Date of Patent: Aug. 6, 1996**

[54] **METHOD OF SIMULATING A ROOM AND/OR SOUND IMPRESSION**

[75] Inventor: **Martin Opitz**, Vienna, Austria

[73] Assignee: **AKG Akustische U. Kino-Geräte Gesellschaft m.b.H.**, Wien, Germany

[21] Appl. No.: **293,134**

[22] Filed: **Aug. 19, 1994**

[30] **Foreign Application Priority Data**

Aug. 26, 1993 [GB] United Kingdom ............. 43 28 620.8

[51] **Int. Cl.**[6] ................................ **H03G 3/00**; H04S 1/00

[52] **U.S. Cl.** ................................................ **381/63**; 381/61

[58] **Field of Search** ............................ 381/61–64, 17–18

[56] **References Cited**

U.S. PATENT DOCUMENTS

| | | | |
|---|---|---|---|
| 5,123,050 | 6/1992 | Serikawa et al. | 381/61 |
| 5,131,051 | 6/1992 | Kishinaga et al. | 381/64 |
| 5,142,586 | 8/1992 | Berkhout | 381/63 |
| 5,201,005 | 4/1993 | Matsushita et al. | 381/63 |
| 5,261,005 | 11/1993 | Masayuki | 381/63 |
| 5,305,386 | 4/1994 | Yamato | 381/63 |
| 5,381,482 | 1/1995 | Matsumoto et al. | 381/63 |

FOREIGN PATENT DOCUMENTS

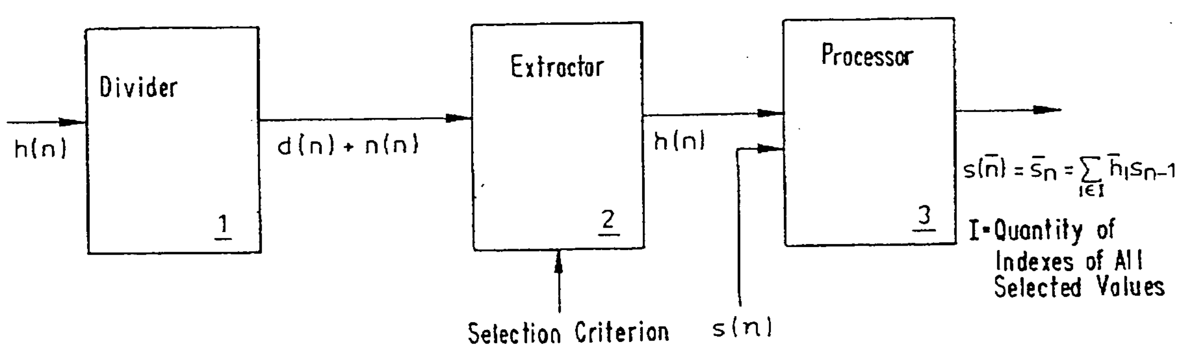
| | | |
|---|---|---|
| 394650 | 11/1990 | Austria. |
| 0505949 | 9/1992 | European Pat. Off.. |

*Primary Examiner*—Stephen Brinich
*Attorney, Agent, or Firm*—Friedrich Kueffner

[57] **ABSTRACT**

A method of simulating a room impression and/or sound impression occurring at a representative listening location in a room with monophonic, stereophonic or multichannel reproduction includes selecting a room whose sound is to be simulated. A location of a representative listening location is then determined. Subsequently, the corresponding room impulse response at least for one channel is determined at the representative listening location. A threshold value which exceeds over at least a portion of the duration of the determined room impulse response is determined for the determined room impulse response. By comparing the determined room impulse response with the threshold value, a reduced room impulse response is produced which within the portion of the duration of the determined room impulse response only includes those contents of the determined room impulse response in which a momentary amplitude is above the threshold value. The reduced impulse response to the value zero for those contents of the determined room impulse response whose momentary amplitude is below the threshold value is set. Outside of the portion of the duration of the determined room impulse response, the reduced room impulse response contains the determined room impulse response in unchanged form.

**14 Claims, 10 Drawing Sheets**



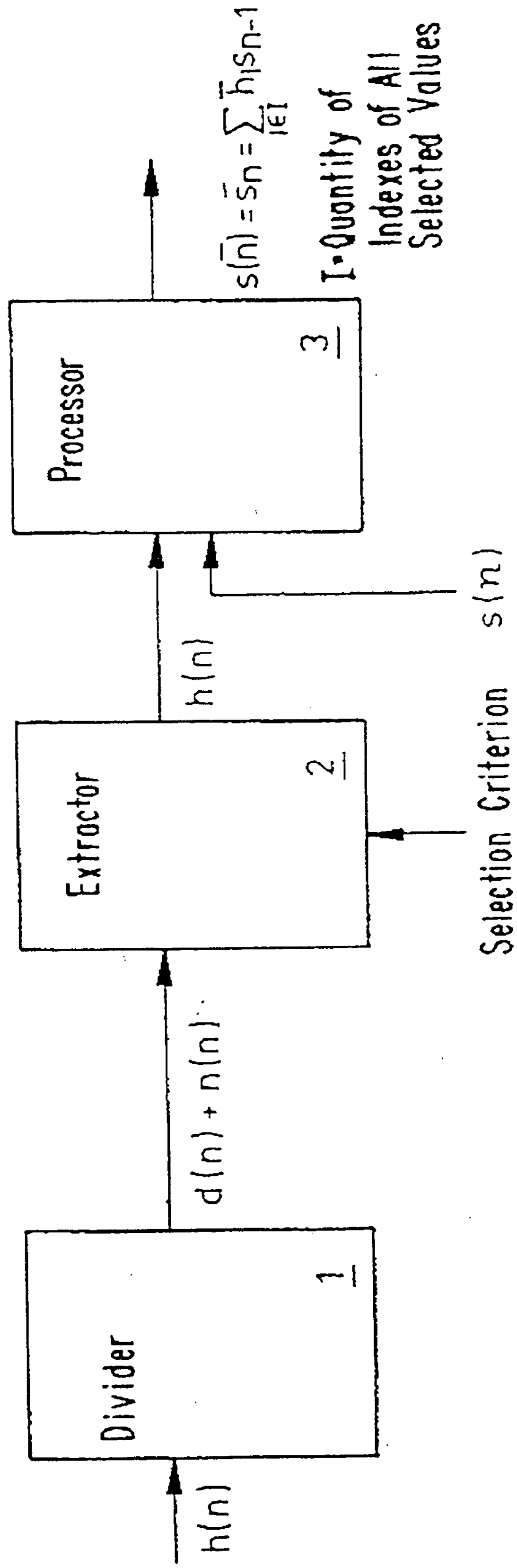
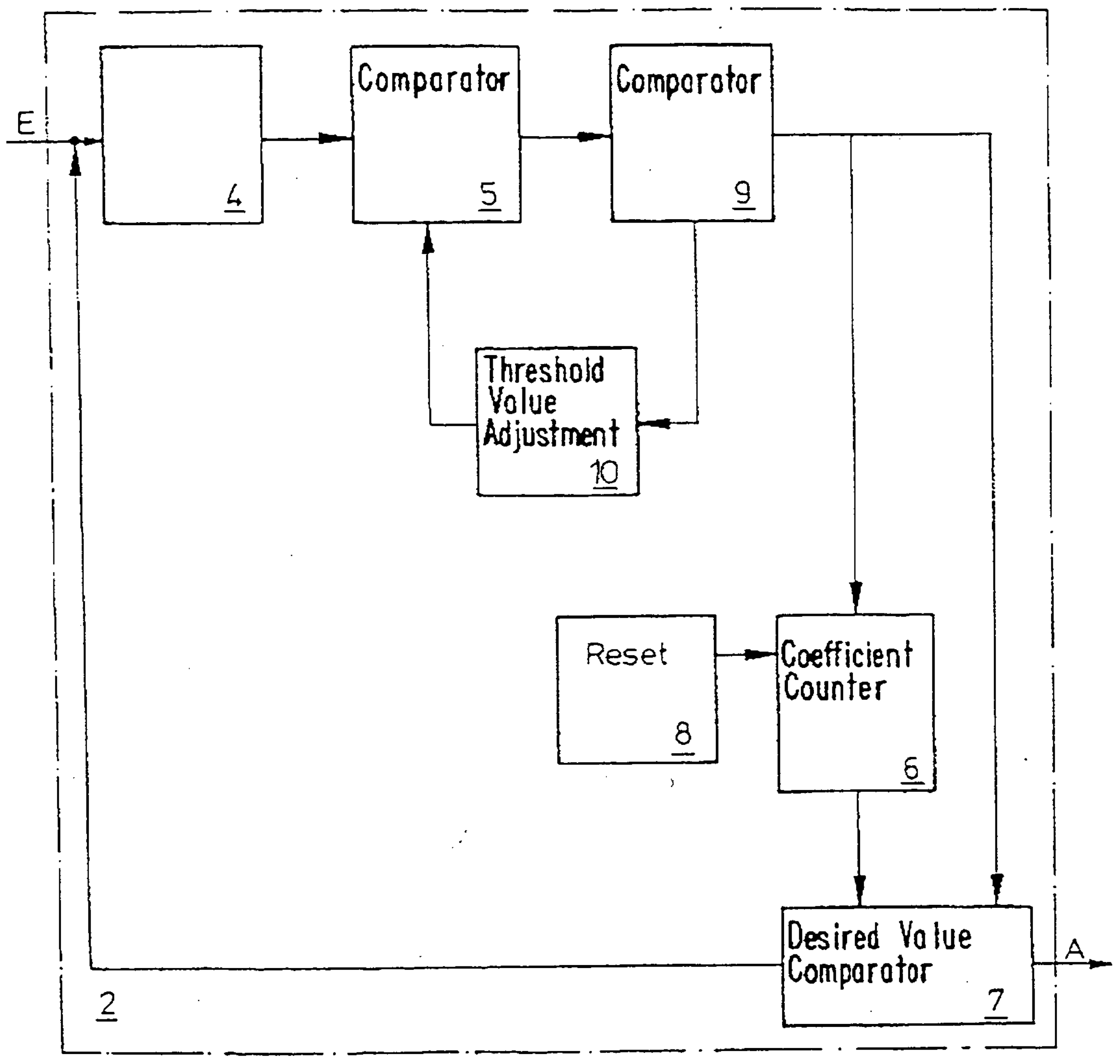

h(n)
Divider
1
d(n) + n(n)
Extractor
2
Selection Criterion
h(n)
s(n)
Processor
3
$s(\bar{n}) = \bar{s}_n = \sum_{l \in I} \bar{h}_l s_{n-1}$
I • Quantity of Indexes of All Selected Values

Fig. 1b



E
4
Comparator
5
Comparator
9
Threshold Value Adjustment
10
Reset
8
Coefficient Counter
6
Desired Value Comparator
7
A
2


Fig. 3

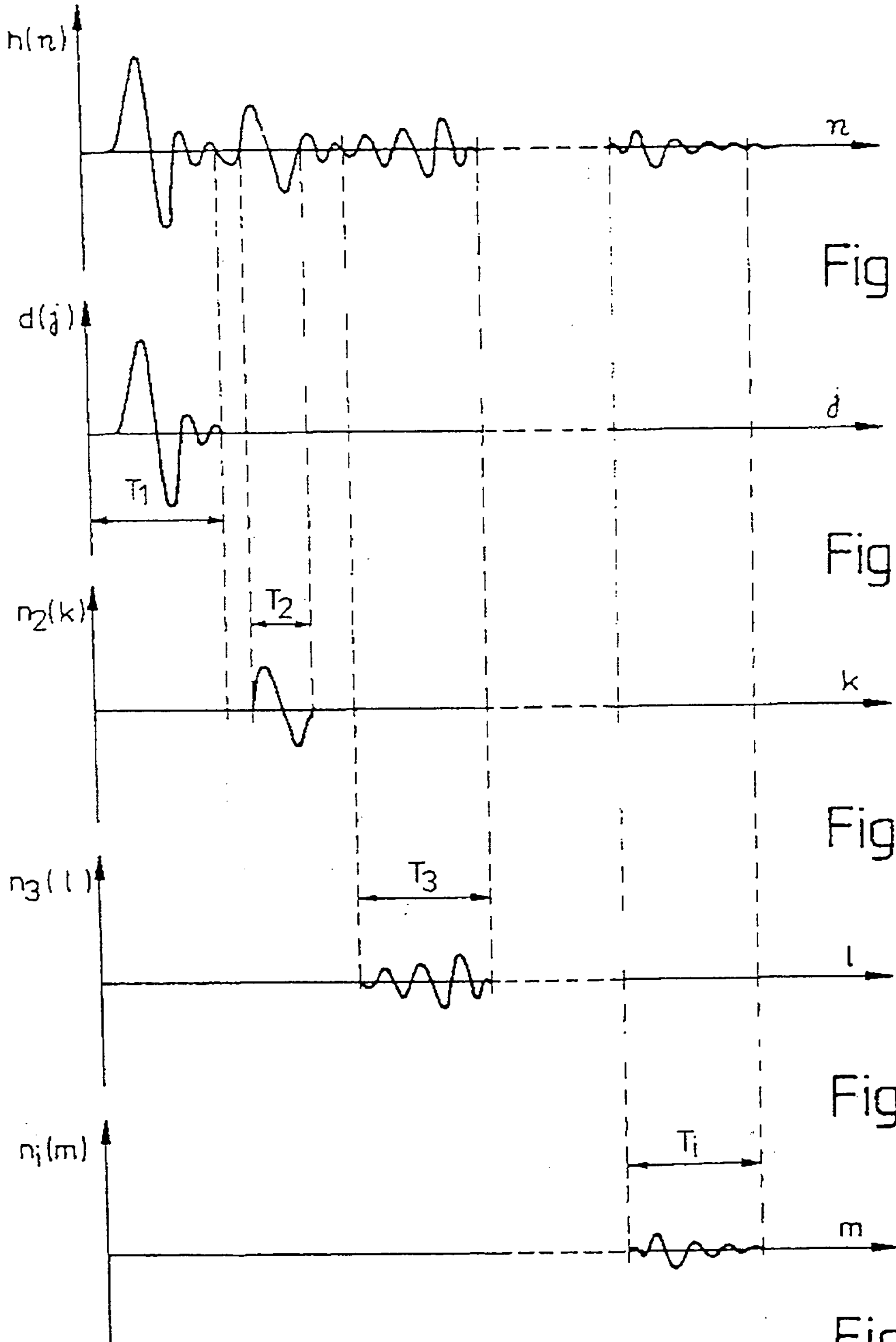



Fig. 5a

Fig. 5b

Fig. 5c

Fig. 5d

Fig. 5e

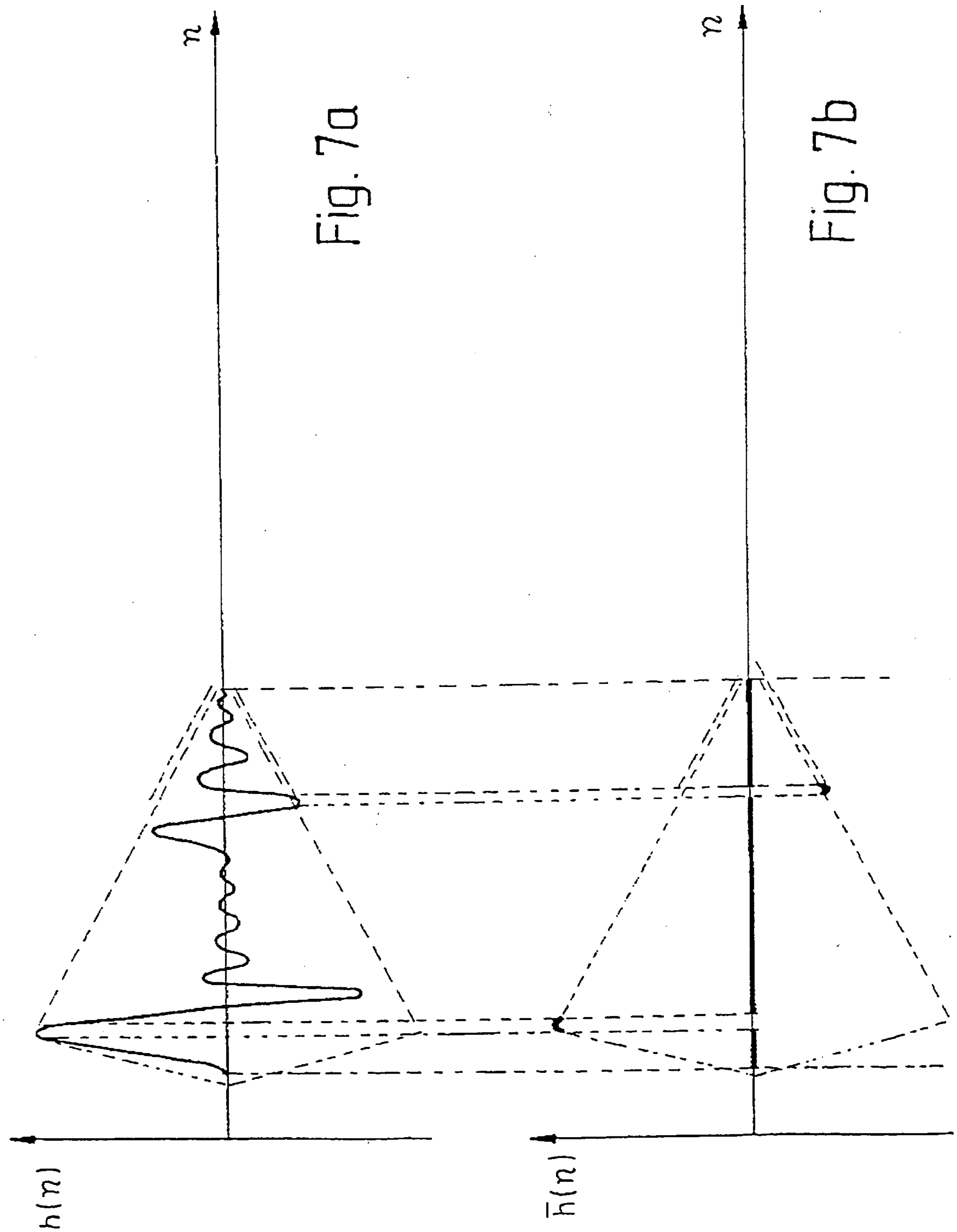



Fig. 7a

Fig. 7b

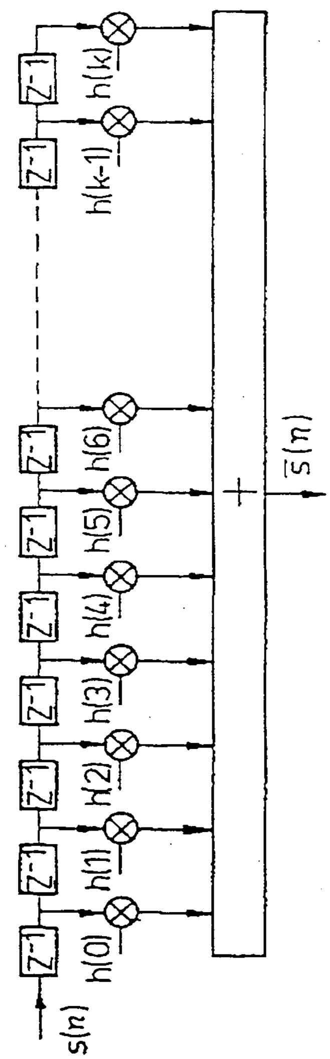

s(n)
Z-1
h(0)
h(1)
h(2)
h(3)
h(4)
h(5)
h(6)
h(k-1)
h(k)
+
s̄(n)

Fig. 9

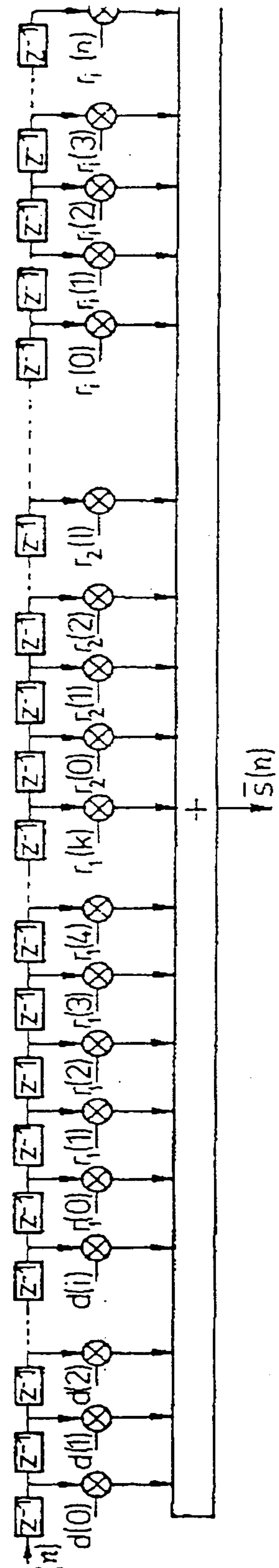

(n)
Z-1
d(0)
d(1)
d(2)
d(i)
r1(0)
r1(1)
r1(2)
r1(3)
r1(4)
r1(k)
r2(0)
r2(1)
r2(2)
r2(l)
ri(0)
ri(1)
ri(2)
ri(3)
ri(n)
+
s̄(n)

Fig. 10

cific, so that for the useful and economical use of this method there is no particular interest.

The known fast convolution by means of discrete Fourier transformation also does not offer a suitable solution for an economical unit for the simulation of room-acoustic events. This is because of the time delay between source signal and convolved signal which is inherent to this method.

## SUMMARY OF THE INVENTION

Therefore, it is the primary object of the present invention to provide a simulation method with the electroacoustic apparatus required for this purpose, which is simplified as compared to known methods, so that the realization of the method is technically and economically feasible.

In accordance with the present invention, the above object is met by a method which includes the steps of:

selecting a room whose sound is to be simulated;

determining within the room the location of a representative listening location;

determining at the representative listening location the corresponding room impulse response at least for one channel;

determining for the determined room impulse response a threshold value which extends over at least a portion of the duration of the determined room impulse response; and

by comparing the determined room impulse response to the threshold value, producing a reduced room impulse response which within the portion of the duration of the determined room impulse response only includes those contents of the determined room impulse response in which the momentary amplitude is above the threshold value, while setting the reduced room impulse response to the value zero for those portions of the determined room impulse response whose momentary amplitude is below the threshold value, and which outside of the portion of the duration of the determined room impulse response contains the determined room impulse response in unchanged form.

Because the method according to the present invention selects certain portions from the room impulse responses, the volume of calculations is reduced accordingly since no calculations must be carried out for the omitted portions of the room impulse responses.

The novel simulation method has the advantage that the simulation quality is not reduced even though necessary computational power is severely reduced. In addition, simplified FIR filter structures can be used for convolution. The convolution process takes place without detectable time delay in real time.

Accordingly, the gist of the present invention resides in that a successful true simulation can be carried out with certain portions of the room impulse responses. It is merely necessary to know those portions of the room impulse responses which in accordance with a critical selection are essential for the auditory impression. The knowledge concerning the respective room impulse responses can be obtained by real room-acoustic measurements or model calculation of existing or virtual rooms. The decision concerning which portions are omitted from the room impulse response is made in accordance with auditory psychological principles.

A significant embodiment of the method according to the present invention provides for comparing the values of the room impulse response with a time-dependent threshold value and using only those values of the room impulse responses which exceed the threshold value. Relative to the room impulse response, the threshold value is time-dependent since it has its greatest value in the range of the beginning of the room impulse response and dies down toward the end of the room impulse response. Consequently, significant portions of the room impulse responses become zero.

The advantage of such a division is the fact that the calculation effort for the simulation processor is significantly reduced. The portion of the room impulse response including the direct sound must be combined with the portion containing the reverberation in such a way that the original quality is maintained in the simulation.

In that manner, only those portions are used for the convolution process which contribute significantly to the true simulation. All other portions of the room impulse response no longer appear as a result of being set to zero and no calculations are required for these portions. The FIR filter used for convolution does not have to have a complicated structure and the computational power of the signal processor does only have to be used when coefficients appear which differ from zero. This procedure reduces the calculation effort significantly as compared to conventional convolution and reduction factors of between 10 and 100 can be achieved. Nevertheless, the reverberation time is maintained for room-acoustic events simulated in this manner; with a total duration of the reduced impulse response of only 10 milliseconds, reverberation times which are between 100 to 1,000 milliseconds are simulated without problems. The spatial simulation is not subject to coincidence.

The above-described method, and the electroacoustic apparatus for carrying out the method, can also be configured in such a way that the critical selection of significant portions for maintaining the true simulation is effected by taking into consideration the psychoacoustic forward-masking and backward-masking phenomena in the room impulse response. The masking phenomena known in acoustics have the effect that in the presence of sound, another second sound can only be heard if its excitation in the human ear exceeds that of the first sound. This creates a displacement of the audibility threshold which is imitated by the above-described time-dependent threshold value, so that sound below this threshold is not perceived.

The combination of the two method sequences mentioned and described above is the optimum embodiment of the method according to the present invention. The yield is the greatest possible in relation to the calculation effort and the use of technical equipment, and the obtained result is the most economical.

The simulation method according to the invention will be used particularly in the fields of Hi-Fi recordings and sound studios because that is where the advantages of binaural listening are for the headset reproduction as well as for loudspeaker reproduction. The apparatus according to the invention provides that degree of good and true room acoustics which cancels out the known disadvantages of listening in an anechoic chamber, while not harmfully superimposing the acoustics provided by the recording. The simulation of, for example, a certain loudspeaker arrangement in a certain room by means of headset reproduction is a significant use of the simulation method and of the electroacoustic apparatus required for carrying out the method.

The various features of novelty which characterize the invention are pointed out with particularity in the claims annexed to and forming a part of the disclosure. For a better

electronic device **2** which extracts from the determined room impulse response the components which contain those characteristics of the listening room acoustics, of the sound field present in the listening room and the left and right outer ear transfer functions assignable to the listener, which after the convolution process with any chosen audio program guarantee the true simulation of the entire room-acoustic event. The extraction is carried out in accordance with criteria which are described further below. The extracted or reduced room impulse response h'(n) is convolved in a processor **3** with the signal s (n) of any selected audio program in order to form the signal. When the sound reproduction is correct at both ears of the listener, the listening result desired in accordance with the invention is achieved, i.e., the true simulation of a listening location in a certain listening room.

The extractor circuit **2** for selecting the significant components from the determined room impulse response is explained in more detail by the diagram of FIG. **2**.

Because of the limited computational capacity of processor **3**, it is advantageous to use only an early part of the respectively determined room impulse response. For this purpose, the room impulse response existing at an input E and divided into the components direct sound and reverberation sound is divided in a function block **4** into individual portions having the duration $T_i$.

FIGS. *5a–5e* show how the determined room impulse response is divided by means of the function block **4** into individual blocks or portions $T_i$ having the sound components $d(n)$, $r_2(n)$, $r_3(n)$ . . . $r_i(n)$.

The division into direct sound and reverberation sound is carried out because the direct component of the determined room impulse response should remain unchanged at least in studio applications and on the reverberation component is reduced as described. However, applications are conceivable in which both components of the determined room impulse response are reduced.

After the direct sound has been separated off, the remaining contents of the room impulse response, which in accordance with a criterion described below are below a predetermined threshold value, are set to zero by means of a comparator **5**. The number of samples in the remaining signal components of the reduced room impulse response are counted in a coefficient counter **6**. The obtained counter value is compared in a desired value comparator **7** to a limit value which is determined by the permissible computing effort. If the limit has not yet been exceeded, additional blocks of the determined room pulse response are called up in accordance with FIGS. *5a–5e*. In this manner, the computing capacity is fully utilized in the case of a later convolution with the reduced room impulse response. When the predetermined desired value has been reached, the now existing reduced room impulse response is conducted to an output A.

In the event that the critical signal evaluation of the determined room impulse response is carried out in accordance with a masking phenomenon, the arrangement illustrated in FIG. **3** is required for this purpose. Compared to the diagram shown in FIG. **2**, a dynamic threshold value adjustment is added in FIG. **3**. The dynamic threshold value adjustment is composed of a comparator **9** and a threshold value generator **10**. In the comparator **9**, the instantaneous value of the determined room impulse response is compared to the instantaneous threshold value, wherein the magnitude of the threshold value is dependent on the preceding values of the determined room impulse response in accordance with the masking phenomenon. Through the return via the threshold value generator **10** to the comparator 5, the dynamic adjustment is realized to the predetermined psychoacoustic criteria in accordance with the masking phenomenon, for example, in accordance with Zwicker.

As illustrated in FIGS. *6a* and *6b*, the critical selection of the signal contents of the determined room impulse response essential for the simulation can be effected by setting to zero all those contents of the determined room impulse response which are below a predetermined fixed threshold value A, so that these contents are not taken into consideration in the later convolution process, while the signal contents exceeding the threshold value are included with unchanged amplitude in the reduced room impulse response. Since there is a direct relationship between the intensity of the sound reflections and the samples of the determined room impulse response corresponding to these reflections, the threshold value criterion constitutes a significant aid in extracting the samples of the determined room impulse response which are essential for the simulation. When convolution is carried out, only the essential features resulting from the selection criterion are taken into consideration from the determined room impulse response, so that the necessary computing effort is substantially reduced. While $25\times10^6$ multiplications and additions can be carried out by the signal processor in the case of a FIR-filter, which corresponds in the case of a sampling interval of 20 μsec to 500 filter coefficients and 10 millisecond impulse response duration, the use of the reduced room impulse response enables the processor to simulate three rooms simultaneously, wherein the reverbation times are up to 1 second.

Finally, as illustrated in FIGS. *7a* and *7b*, the critical selection can also be carried out pursuant to criteria in accordance with masking phenomena. In accordance with these phenomena, those contents of the determined room impulse response do not have to be taken into consideration which are not perceivable during listening anyway. In accordance with the information which is present, the masked contents are to be excluded from the convolution process which is carried out later. In that case, it is also no longer necessary to distinguish between direct sound and reverberation component rather, the entire determined room impulse response can be reduced from the beginning as described above.

$T_v$ designates the areas of forward-masking and $T_N$ designates the areas of backward-masking. These are the periods in which signals below a level limit, as they are sketched in FIG. *7a*, are no longer perceivable compared with the principal signal. As described in the standard literature concerning this topic, the masking effects are dependent on the time spacing, on the level ratio and the frequency spacing of masked signal and masking signal. Consequently, this cannot be completely illustrated in the drawing. The room impulse response primarily influences the time conditions and level conditions. Accordingly, it is always necessary to use somewhat wider value ranges of the determined room impulse response than would result directly from the boundary line criterion. In addition, in order not to obtain undesirable filter effects in the frequency range, it is necessary to extrapolate value ranges into the actually masking range.

FIGS. *8a* and *8b* illustrate how the threshold value decreases in a step-like manner and how the signal contents are determined for the simulation.

FIG. 9 of the drawing shows the possible architecture of a conventional FIR-filter. In the chain of stack memories $z^{-1}$, each of which stores a signal value for a sampling interval,

the circuit comprising

at least one input for feeding in one of a monophonic, a stereophonic and a multichannel audio program,

at least one channel and for each channel at least one audio output for outputting a Processed audio program obtained by convolving the fed-in audio program with the reduced room impulse response for each channel.

14. The apparatus according to claim 13, comprising for each channel at least one FIR filter having filter coefficients corresponding to amplitude values of the reduced room pulse response which is digitalized with a predetermined sampling frequency.

* * * * *

UNITED STATES PATENT AND TRADEMARK OFFICE

# CERTIFICATE OF CORRECTION

PATENT NO. : 5,544,249

DATED : August 6, 1996

INVENTOR(S) :

Martin Opitz

It is certified that error appears in the above-indentified patent and that said Letters Patent is hereby corrected as shown below:

On the title page, item:

[30] **Foreign Application Priority Data**

Aug. 26, 1993 [DE] Germany .......... 43 28 620.8

Signed and Sealed this

Twenty-eighth Day of January, 1997

Attest:

BRUCE LEHMAN

Attesting Officer

Commissioner of Patents and Trademarks

# UNITED STATES PATENT AND TRADEMARK OFFICE
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PATENT NO. : 5,544,249

DATED : August 6, 1996

INVENTOR(S) : Martin Opitz

It is certified that error appears in the above-indentified patent and that said Letters Patent is hereby corrected as shown below:

On title page, item [73] assignee, should read as follows:

[73] Assignee: **AKG Akustische U. Kino-Geräte Gesellschaft m.b.H.,** <u>Vienna, Austria</u>

Signed and Sealed this

Twenty-second Day of April, 1997

Attest:

Bruce Lehman

BRUCE LEHMAN

*Commissioner of Patents and Trademarks*

*Attesting Officer*